Avaya Solution & Interoperability Test Lab

Application Notes for Configuring Open Text Fax Server, RightFax Edition with Avaya Aura® Communication Manager and Avaya Aura® Session Manager via SIP Trunk Interface - Issue 1.0

Abstract

These Application Notes describe the procedures for configuring the Open Text Fax Server, RightFax Edition with Avaya Aura® Communication Manager and Avaya Aura® Session Manager using a SIP trunk interface.

Open Text Fax Server, RightFax Edition is a software based fax server that sends and receives fax calls over an IP network. In the tested configuration, Open Text Fax Server, RightFax Edition interoperated with Avaya Aura® Session Manager to send/receive faxes using SIP trunk facilities.

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 procedures for configuring Open Text Fax Server, RightFax Edition with Avaya Aura® Communication Manager and Avaya Aura® Session Manager using SIP trunks.

Open Text Fax Server, RightFax Edition is a software based fax server that sends and receives fax calls over an IP network. Open Text Fax Server, RightFax Edition utilizes the Brooktrout SR140 T.38 Fax over Internet Protocol (FoIP) virtual fax board software from Dialogic. In the tested configuration, Open Text Fax Server, RightFax Edition interoperated with Avaya Aura® Session Manager to send/receive faxes using a SIP trunk interface.

1.1. Interoperability Compliance Testing

The compliance test tested interoperability between RightFax and Session Manager by making intra-site and inter-site fax calls to and from a RightFax server that was connected (at each of the two sites in the test configuration) to Session Manager via SIP interface. Specifically, the following fax operations were tested in the setup for the compliance test:

- Fax from/to RightFax to/from fax machine at a local site
- Fax from/to RightFax to/from fax machine at a remote site
- Fax from/to RightFax to/from RightFax server at a remote site

In the compliance test, both ISDN-PRI trunks and SIP trunks directly connecting two Communication Manager systems connected the Main Site, and Remote Site.

Faxes were sent with various page lengths and resolutions. For capacity, a large number of 2-page faxes were continuously sent between the two RightFax servers. Serviceability testing included verifying proper operation/recovery from failed cables, unavailable resources, restarts of Session Manager as well as RightFax reboots. Fax calls were also tested with different Avaya Media Gateway media resources to process the fax data between locations. This included the TN2302AP IP Media Processor (MedPro) circuit pack and the TN2602AP IP Media Processor circuit pack in the Avaya G650 Media Gateway; the integrated Voice over Internet Protocol (VoIP) engine of the Avaya G450 Media Gateway, and the Avaya MM760 Media Module installed in the Avaya G450 Media Gateway.

1.2. Support

Technical support for RightFax can be obtained by contacting Open Text at

- Phone: (800) 540-7292
- Email: support@opentext.com
- https://cslogin.opentext.com/login/

2. Configuration

The test configuration was designed to emulate two separate sites with multiple Port Networks at one site, and modular Gateway resources at the other site. **Figure 1** illustrates the configuration used in these Application Notes.

2.1. Configuration Details

In the sample configuration, 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. Connections to Session Manager were via SIP trunk facilities, and the RightFax servers communicated directly with Session Manager via SIP.

Two separate Session Manager Servers were used to connect the RightFax Servers to each site.

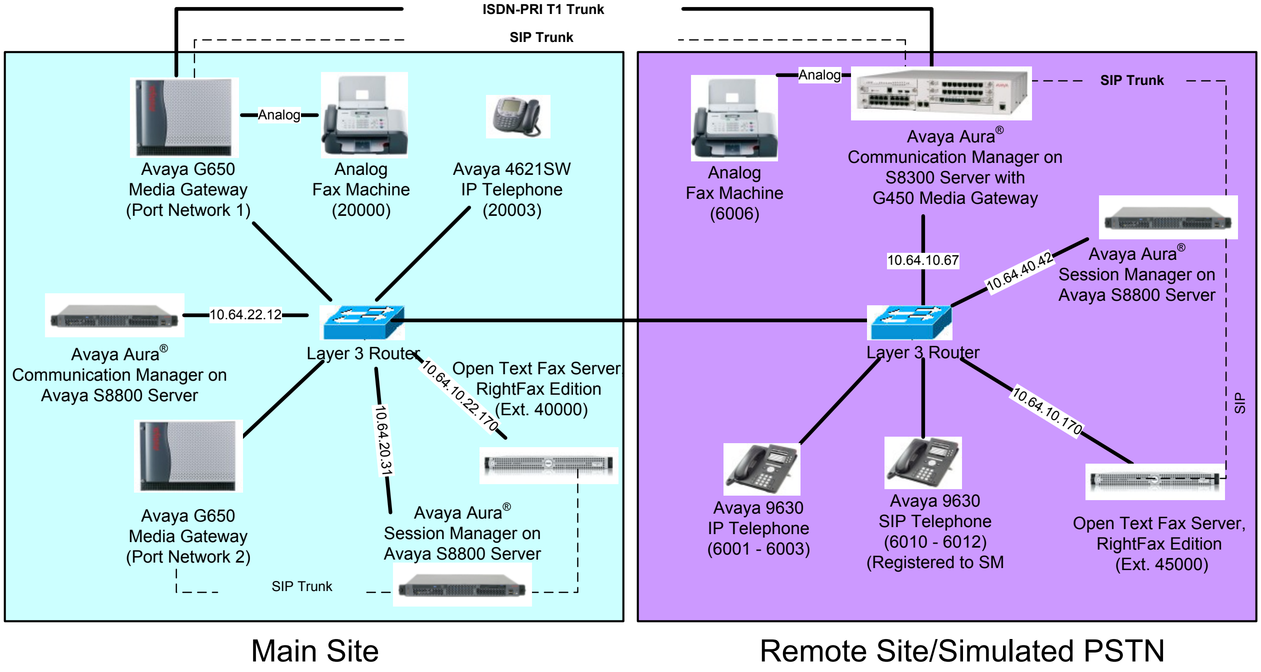


Figure 1: RightFax interoperating with Session Manager via SIP Trunk

The Main Site 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 RightFax server at this site communicated with Session Manager via SIP. In turn, Communication Manager used a SIP Trunk which terminated on a CLAN circuit pack in port network 2 to communicate with Session Manager. 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 4600 Series IP Telephone (with H.323 firmware), and an analog fax machine.

The Remote Site had an Avaya S8300 Server running Communication Manager in an Avaya G450 Media Gateway. The RightFax server at this site communicated with Session Manager via SIP. On the Avaya G450 Media Gateway, the signaling and media resources supporting a SIP trunk connected to Session Manager were integrated directly on the media gateway processor. Endpoints at this site included Avaya 9600 Series IP Telephones (with H.323 and SIP firmware), and an analog fax machine.

The IP telephones were not involved in the faxing operations, they were present in the configuration to verify the effect VoIP telephone calls had on the FoIP faxing operations.

Outbound fax calls originating from RightFax were sent to Session Manager first, then to Communication Manager, via the configured SIP trunks. Based on the dialed digits, Communication Manager directed the calls to the local fax machine, or the inter-site trunks (ISDN-PRI or SIP) to reach the Remote Site. Inbound fax calls to RightFax were first received by Communication Manager from the local fax machine or from across either ISDN-PRI or SIP trunks connected to the Remote Site. Communication Manager then directed the calls to RightFax via the configured Session Manager SIP trunks.

3. Equipment and Software Validated

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

Equipment	Software/Firmware
Avaya S8800 Servers (at both sites)	Avaya Aura® Session Manager 6.0 Avaya Aura® System Manager 6.0
Avaya S8800 Server (at Main Site)	Avaya Aura® Communication Manager 6.0 SP1 R016x.00.0.345.0 with patch 18567
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 S8300D Server (at Remote Site)	Avaya Aura® Communication Manager 6.0 SP1 R016x.00.0.345.0 with patch 18567
Avaya G450 Media Gateway (at Remote Site)	30.14.0/1
Avaya 9620 IP Telephone (SIP) Avaya 9630 IP Telephone (H.323) Avaya 4621SW IP Telephone (H.323)	Avaya one-X® Deskphone Edition SIP 2.5 H.323 3.11 H.323 2.9
Analog Fax Machines	-
Open Text Fax Server, RightFax Edition on Windows Server 2008R2	9.4 Feature Pack 1 Service Release 3
Dialogic Brooktrout SR140 Fax Software – Boston Bfv API – Boston Driver – Boston SDK – Boot ROM	 v6.2.04 (Build 12) v6.2.00 (Build 4) v6.2.04 (Build 12) 6.2.1B9

4. Configure Avaya Aura® Communication Manager

This section describes the Communication Manager configuration necessary to interoperate with Session Manager and Open Text Fax Server, RightFax Edition. It focuses on the configuration of the SIP trunks connecting Communication Manager to the Avaya SIP infrastructure with the following assumptions:

- The examples shown in this section refer to the Main Site. Unless specified otherwise, these same steps also apply to the Remote Site 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.

4.1. Steps to Configure Communication Manager

The procedures for configuring Communication Manager include the following areas:

- Verify Communication Manager License (Step 1)
- Identify IP Interfaces (Step 2)
- Administer IP Network Regions (Steps 3 – 6)
- Administer IP Node Name (Step 7)
- Administer IP Network Map (Step 8)
- Administer IP Codec Set (Steps 9 – 10)
- Administer SIP Signaling Group (Step 11)
- Administer SIP Trunk Group (Steps 12 – 13)
- Administer Public Unknown Numbering (Step 14)
- Administer Route Pattern (Step 15)
- Administer AAR Analysis (Steps 16 – 17)

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 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: 12000 96
           Maximum Concurrently Registered IP Stations: 18000 1
             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 H.323 Stations: 100   0
                        Maximum Video Capable Stations: 18000 0
                   Maximum Video Capable IP Softphones: 18000 0
                       Maximum Administered SIP Trunks: 24000 298
  Maximum Administered Ad-hoc Video Conferencing Ports: 24000 0
   Maximum Number of DS1 Boards with Echo Cancellation: 522   0
                             Maximum TN2501 VAL Boards: 128   2
                     Maximum Media Gateway VAL Sources: 250   0
           Maximum TN2602 Boards with 80 VoIP Channels: 128   0
          Maximum TN2602 Boards with 320 VoIP Channels: 128   2
   Maximum Number of Expanded Meet-me Conference Ports: 300   0
```

Step	Description
2.	**Identify IP Interfaces** Use the **list ip-interface clan** and **list ip-interface medpro** commands to identify IP interfaces in each network region. Interfaces in cabinet 01 (port network 1) as indicated in the **Slot** field are in IP network region 1 as indicated in the **Net Rgn** field. Testing with the TN2302 and TN2602 circuit packs were done separately. When testing with the TN2302, the TN2602 was disabled (turned off) and vice versa as indicated in the **ON** field.

```
list ip-interface clan

                                IP INTERFACES

                                                            Skts  Net       Eth
ON  Slot   Code/Sfx Node Name/       Mask  Gateway Node     Warn  Rgn VLAN  Link
                    IP-Address
--  ----   -------- ---------------  ----  ---------------  ----  --- ---- ----
 y  01A03  TN799  D CLAN1A           /24   Gateway001       400   1   n    1
                    10.64.22.16
 y  02A03  TN799  D CLAN2A           /24   Gateway001       400   2   n    2
                    10.64.22.19
```

```
list ip-interface medpro

                                IP INTERFACES

                                                       Net
ON Slot  Code/Sfx Node Name/      Mask Gateway Node    Rgn VLAN Virtual Node
                  IP-Address
-- ----- -------- --------------- ---- --------------- --- ---- ---------------
 n 01A02 TN2302   MEDPRO1A        /24  Gateway001      1   n
                  10.64.22.15
 n 02A02 TN2302   MEDPRO2A        /24  Gateway001      2   n
                  10.64.22.18
 y 01A04 TN2602   MEDPRO1A-2      /24  Gateway001      1   n
                  10.64.22.17
 y 02A04 TN2602   MEDPRO2A-2      /24  Gateway001      2   n
                  10.64.22.20
```

Step	Description
3.	**Administer IP Network Region 1**

The configuration of the IP network regions (**Steps 3 – 6**) was already in place and is included here for clarity. At the Main Site, the Avaya G650 Media Gateway comprising port network 1 and all IP endpoints were located in IP network region 1.

Use the **display ip-network-region** command to view these settings.

A descriptive name was entered for the **Name** field.

- **IP-IP Direct Audio** (Media Shuffling) was enabled to allow audio traffic to be sent directly between IP endpoints without using media resources in the Avaya Media Gateway. This was done for both intra-region and inter-region IP-IP Direct Audio. This is the default setting. Media Shuffling can be further restricted at the trunk level on the **Signaling Group** form.
- The **Codec Set** field was set to the IP codec set to be used for calls within this IP network region. In this case, IP codec set 1 was selected.
- The default values were used for all other fields.

At the Remote Site, all IP components were located in IP network region 1 and the IP network region was configured in the same manner as shown below.

```
display ip-network-region 1                   Page  1 of 20
               IP NETWORK REGION
 Region: 1
Location:     Authoritative Domain: avaya.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
```

Step	Description
4.	**Administer IP Network Region 1 – Continued** On **Page 4**, codec sets are defined for inter-region calls. In the case of the compliance test at the Main Site, calls from IP network Source Region 1 to IP network region 2 (**dst rgn 2**) used codec set 1. The default values were used for all other fields. At the Remote Site, only one IP network region was used, so no inter-region settings were required.

```
display ip-network-region 1                                     Page   4 of  20

 Source Region: 1     Inter Network Region Connection Management     I       M
                                                                     G  A    t
 dst codec direct   WAN-BW-limits   Video      Intervening      Dyn  A  G    c
 rgn  set   WAN  Units    Total Norm  Prio Shr Regions          CAC  R  L    e
 1    1                                                                all
 2    1     y    NoLimit                                             n       t
 3
```

Step	Description
5.	**Administer IP Network Region 2** At the Main Site, IP network region 2 was created for Port Network 2 in a similar manner as IP network region 1 shown in **Step 3** but with a different name. This was the network region used for H.323 Trunk connections to the RightFax server. Note the default **UDP Port Min** and **UDP Port Max** settings, these were used in the configuration of the RightFax server described in **Section 5**.

```
display ip-network-region 2                                     Page   1 of  20
                               IP NETWORK REGION
  Region: 2
Location:         Authoritative Domain: avaya.com
    Name: PN2
MEDIA PARAMETERS                Intra-region IP-IP Direct Audio: yes
      Codec Set: 1              Inter-region IP-IP Direct Audio: yes
   UDP Port Min: 2048                      IP Audio Hairpinning? n
   UDP Port Max: 3329
DIFFSERV/TOS PARAMETERS
 Call Control PHB Value: 46
        Audio PHB Value: 46
        Video PHB Value: 26
802.1P/Q PARAMETERS
 Call Control 802.1p Priority: 6
        Audio 802.1p Priority: 6
        Video 802.1p Priority: 5      AUDIO RESOURCE RESERVATION PARAMETERS
H.323 IP ENDPOINTS                                       RSVP Enabled? n
  H.323 Link Bounce Recovery? y
 Idle Traffic Interval (sec): 20
   Keep-Alive Interval (sec): 5
            Keep-Alive Count: 5
```

Step	Description
6.	**Administer IP Network Region 2 – Continued** The inter-region codec setting was created similarly to **Step 4**.

```
display 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
 3    3     y    NoLimit                                             n all
```

Step	Description
7.	**Administer IP Node Name** Use the **change node-names ip** command to create a node name that maps to the Session Manager IP address. This node name is used in the configuration of the SIP trunk signaling group in **Step 11**.

```
change node-names ip                                            Page   1 of   2
                              IP NODE NAMES
    Name              IP Address
CLAN1A              10.64.22.16
CLAN2A              10.64.22.19
CM-Remote           10.64.21.111
DemoSM              10.64.20.31
Gateway001          10.64.22.1
MEDPRO1A            10.64.22.15
MEDPRO1A-2          10.64.22.17
MEDPRO2A            10.64.22.18
MEDPRO2A-2          10.64.22.20
TR18300             10.64.10.67
```

Step	Description
8.	**Administer IP Network Map** Session Manager and the RightFax server were configured to be located in an IP network region other than the default region 1; the region was assigned using the **change ip-network-map** command. In the case of the compliance test, the IP addresses for these resources at the Main Site were assigned to IP network region 2 as shown in the example below. At the Remote Site, Session Manager and the RightFax server were located in the default IP network region 1, so it did not require an IP address map entry.

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

                                                Subnet Network     Emergency
 IP Address                                     Bits   Region VLAN Location Ext
 ---------------------------------------------- ------ ------ ---- -------------
 FROM: 10.64.20.31                               /      2      n
   TO: 10.64.20.31
 FROM: 10.64.22.170                              /      2      n
   TO: 10.64.22.170
```

Step	Description
9.	**Administer IP Codec set** Use the **change ip-codec-set 1** command to verify that G.711MU or G.711A is contained in the codec list. The example below shows the value used in the compliance test.

```
display 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
```

Step	Description
10.	**Administer IP Codec set – Fax settings** On **Page 2**, set the **FAX Mode** field to ***t.38-standard*.** This is necessary to support the RightFax server assigned to IP network region 2. 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. This setting should match the redundancy settings in Brooktrout SR140 configuration; otherwise Brooktrout SR140 will negotiate T.38 redundancy to the most common denominator (no redundancy in this case).

```
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
```

Step	Description
11.	**Administer SIP Signaling Group**

For the compliance test, a signaling group and the associated SIP trunk group was used for routing fax calls to/from the RightFax server via Session Manager. For the compliance test at the Main Site, signaling group 12 was configured using the parameters highlighted below. All other fields were set as described in [3].

- The **Group Type** was set 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.***
- The **Near-end Node Name** was set to ***CLAN2A***, the node name that maps to the IP address of the CLAN circuit pack used to connect to Session Manager. Node names are defined using the **change node-names ip** command (see **Step 7** above).
- The **Far-end Node Name** was set to ***demoSM***. This node name maps to the IP address of the Session Manager server as defined using the **change node-names ip** command.
- The **Far-end Network Region** was set to ***2***. This is the IP network region which contains Session Manager and RightFax.
- The **Far-end Domain** was set to ***avaya.com***. This domain is sent in the headers of SIP INVITE messages for calls originating from and terminating to Session Manager using this signaling group.
- **Direct IP-IP Audio Connections** was set to ***y***. This field must be set to ***y*** to enable Media Shuffling on the trunk level (see **Step 3** on **IP-IP Direct Audio**).
- The **DTMF over IP** field was set to the default value of ***in-band***.
- The default values were used for all other fields.

```
change signaling-group 12                                       Page   1 of   1
                                SIGNALING GROUP

 Group Number: 12              Group Type: sip
  IMS Enabled? n         Transport Method: tls
        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: CLAN2A               Far-end Node Name: demoSM
 Near-end Listen Port: 5061               Far-end Listen Port: 5061
                                       Far-end Network Region: 2

Far-end Domain: avaya.com
                                              Bypass If IP Threshold Exceeded? n
Incoming Dialog Loopbacks: eliminate                   RFC 3389 Comfort Noise? n
         DTMF over IP: in-band                 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
```

Step	Description
12.	**Administer SIP Trunk Group** For the compliance test, trunk group 12 was used for the SIP trunk group for routing fax calls to/from Session Manager. Trunk group 12 was configured using the parameters highlighted below. All other fields were set as described in [3]. On **Page 1**: ▪ The **Group Type** field was set to ***sip***. ▪ A descriptive name was entered for the **Group Name**. ▪ An available trunk access code (TAC) that was consistent with the existing dial plan was entered in the **TAC** field. ▪ The **Service Type** field was set to ***tie***. ▪ The **Signaling Group** was set to the signaling group shown in the previous step. ▪ The **Number of Members** field contained the number of trunks in the SIP trunk group. It determines how many simultaneous SIP calls can be supported by the configuration. Each SIP call between two SIP endpoints (whether internal or external) requires two SIP trunks for the duration of the call. ▪ The default values were used for all other fields.

```
change trunk-group 12                                           Page   1 of  21
                                TRUNK GROUP

Group Number: 12                   Group Type: sip          CDR Reports: y
  Group Name: PN2 to demoSM               COR: 1        TN: 1        TAC: *012
   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: 12
                                                     Number of Members: 50
```

Step **Description**

13. **Administer SIP Trunk Group – continued**

On **Page 3**:

- Set the **Numbering Format** field to ***public***. This field specifies the format of the calling party number sent to the far-end.
- Default values may be used for all other fields.

```
change trunk-group 12                                           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


                                Modify Tandem Calling Number: no




 Show ANSWERED BY on Display? y
```

14. **Administer Public Unknown Numbering**

Public unknown numbering defines the calling party number to be sent to the far-end. Use the **change public-unknown-numbering** command to create an entry that will be used by the trunk groups defined in **Steps 12-13**. In the example shown below, all calls originating from a 5-digit extension beginning with 2 or 4 and routed across any trunk group (**Trk Grp** column is blank) were sent as a 5-digit calling number.

```
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: 6
 5  1                                       5        Maximum Entries: 9999
 5  2                                       5
 5  4                                       5     Note: If an entry applies to
 5  5                                       5     a SIP connection to Avaya
 4  6                                       4     Aura(tm) Session Manager,
 5  7                                       5     the resulting number must
                                                  be a complete E.164 number.
```

Step	Description
15.	**Administer Route Pattern** Use the **change route-pattern** command to create a route pattern that will route fax calls to the SIP trunk that connects to the RightFax server. The example below shows the route pattern used for the compliance test at the Main Site. A descriptive name was entered for the **Pattern Name** field. The **Grp No** field was set to the trunk group created in **Steps 12–13**. 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 default values were used for all other fields.

```
change route-pattern 12                                         Page   1 of   3
                    Pattern Number: 12  Pattern Name: To SM
                             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: 12   0                                                              n   user
```

Step	Description
16.	**Administer AAR Analysis** Automatic Alternate Routing (AAR) was used to route calls to RightFax via Session Manager. Use the **change aar analysis** command to create an entry in the AAR Digit Analysis Table for this purpose. The example below shows entries previously created for the Main Site using the **display aar analysis 0** command. The 3rd highlighted entry specifies that 5 digit dial string 40000 was to use route pattern 12 to route calls to the RightFax fax server at the Main Site via Session Manager. The dial string 45000 (the RightFax server at the Remote Site) used Route Pattern 15 to route calls between Communication Managers.

```
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
    10                    4    4     4         aar          n
    3                     5    5     12        aar          n
    40000                 5    5     12        aar          n
    45000                 5    5     15        aar          n
```

5. Configure Avaya Aura® Session Manager - Overview

This section covers the configuration of Session Manager at the Main Site. Session Manager is configured via an Internet browser using the administration web interface. It is assumed that the setup screens of the administration web interface have been used for initial configurations. For additional information on these installation tasks, refer to [3].

Each SIP endpoint used in the compliance test that registered with Session Manager required that a user and endpoint profile be created and associated with Session Manager. This configuration is not directly related to the interoperability of the products being tested, so it is not included here. These procedures are covered in [3].

This section summarizes the configuration steps that are necessary for interoperating with RightFax. 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 are not described in this document, and additional information can be obtained in [3].

The documented configurations were repeated for the Session Manager at the Remote Site using values appropriate for the Remote Site from **Figure 1**. This includes but is not limited to the IP addresses, SIP domain and user extensions.

The steps used were:

- Create a SIP Entity for the RightFax Server
- Create a SIP Entity Link for the RightFax Server
- Create a Routing Policy
- Create or Modify Dial Patterns

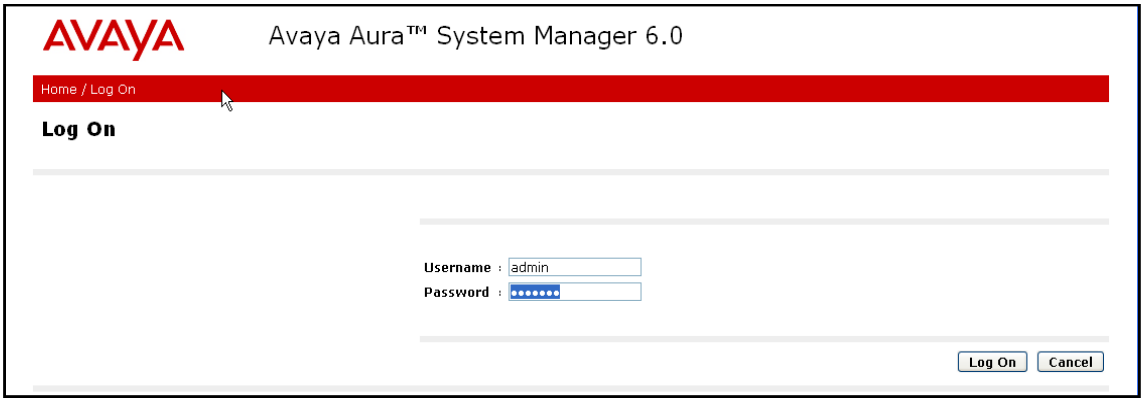
5.1. Configure Session Manager - Details

This section summarizes the applicable user-defined parameters used during the SIP installation procedures.

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

Login
Access the System Manager administration web interface by entering https://<*ip-addr*>/SMGR as the URL in an Internet browser, where <*ip-addr*> is the IP address (or FQDN) of the System Manager server.

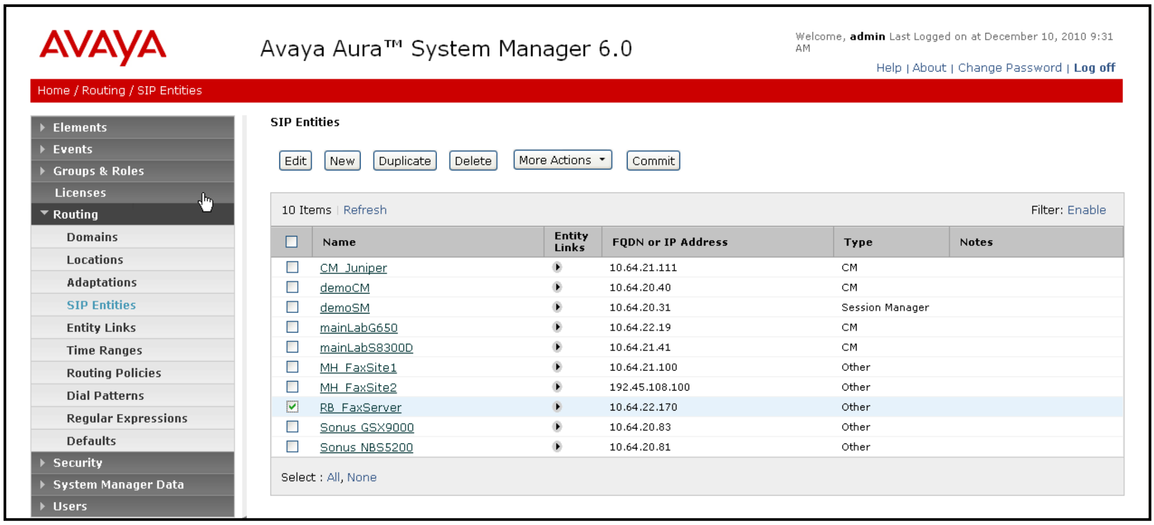
Log in with the appropriate credentials.



Username : admin
Password : •••••••

</td></tr>
<tr><td>2.</td><td>

Create a SIP Entity for the RightFax Server
Navigate to **Routing\SIP Entities** and click **New** to create an Entity definition. In the screenshot below, the Entity ***RB_FaxServer*** was previously created using the following settings.



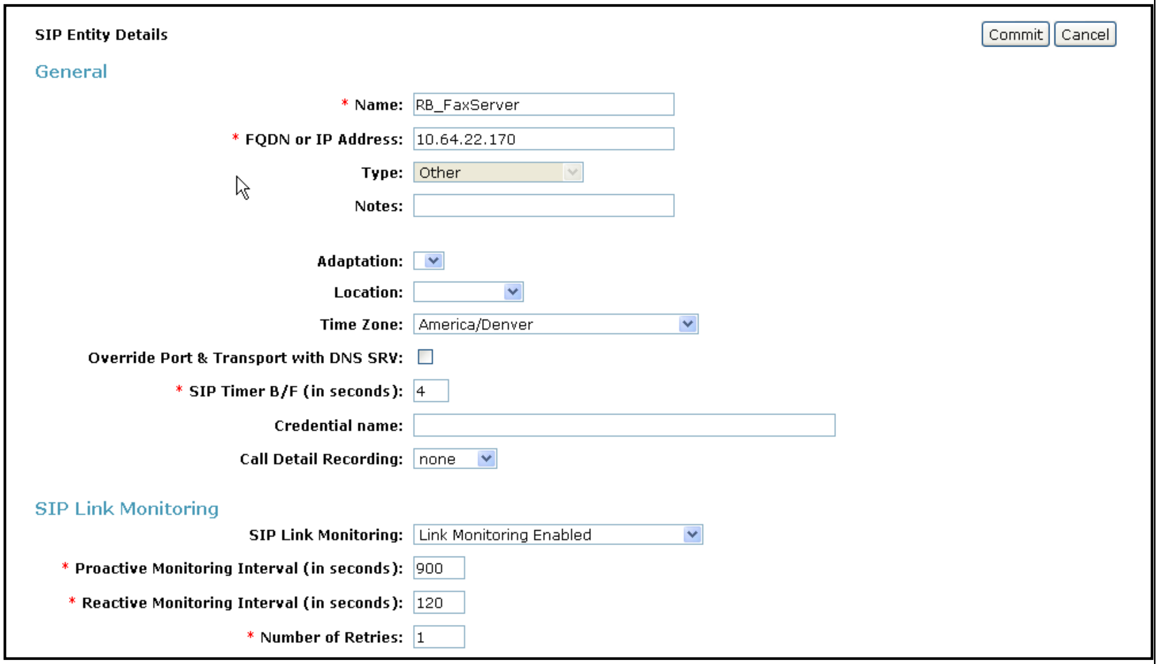
SIP Entities

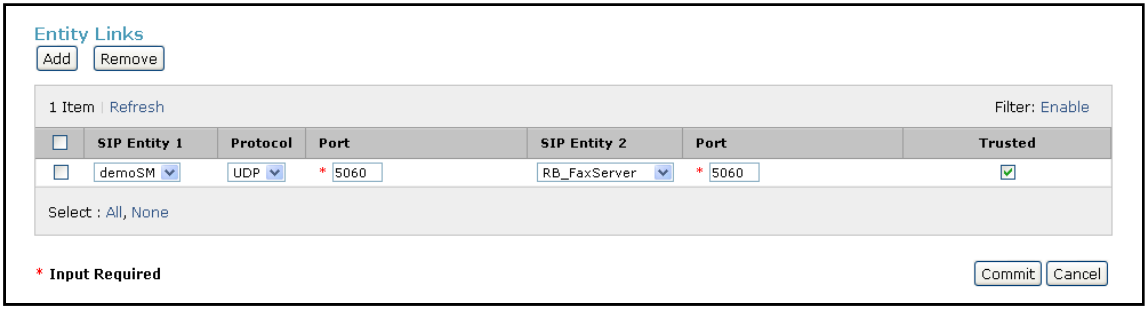
10 Items

<table>
<tr><th>☐</th><th>Name</th><th>Entity Links</th><th>FQDN or IP Address</th><th>Type</th><th>Notes</th></tr>
<tr><td>☐</td><td>CM_Juniper</td><td>▸</td><td>10.64.21.111</td><td>CM</td><td></td></tr>
<tr><td>☐</td><td>demoCM</td><td>▸</td><td>10.64.20.40</td><td>CM</td><td></td></tr>
<tr><td>☐</td><td>demoSM</td><td>▸</td><td>10.64.20.31</td><td>Session Manager</td><td></td></tr>
<tr><td>☐</td><td>mainLabG650</td><td>▸</td><td>10.64.22.19</td><td>CM</td><td></td></tr>
<tr><td>☐</td><td>mainLabS8300D</td><td>▸</td><td>10.64.21.41</td><td>CM</td><td></td></tr>
<tr><td>☐</td><td>MH_FaxSite1</td><td>▸</td><td>10.64.21.100</td><td>Other</td><td></td></tr>
<tr><td>☐</td><td>MH_FaxSite2</td><td>▸</td><td>192.45.108.100</td><td>Other</td><td></td></tr>
<tr><td>☑</td><td>RB_FaxServer</td><td>▸</td><td>10.64.22.170</td><td>Other</td><td></td></tr>
<tr><td>☐</td><td>Sonus_GSX9000</td><td>▸</td><td>10.64.20.83</td><td>Other</td><td></td></tr>
<tr><td>☐</td><td>Sonus_NBS5200</td><td>▸</td><td>10.64.20.81</td><td>Other</td><td></td></tr>
</table>

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

<table>
<tr><th>Step</th><th>Description</th></tr>
<tr><td>3.</td><td>Create a SIP Entity for the RightFax Server - Continued
Enter a descriptive Name such as <i>RB_FaxServer</i> and enter the FQDN or IP Address for the RightFax server as shown below. Select Other for the Entity Type. All other settings were defaults.


SIP Entity Details
General
* Name: RB_FaxServer
* FQDN or IP Address: 10.64.22.170
Type: Other
Notes:
Adaptation:
Location:
Time Zone: America/Denver
Override Port & Transport with DNS SRV: ☐
* SIP Timer B/F (in seconds): 4
Credential name:
Call Detail Recording: none
SIP Link Monitoring
SIP Link Monitoring: Link Monitoring Enabled
* Proactive Monitoring Interval (in seconds): 900
* Reactive Monitoring Interval (in seconds): 120
* Number of Retries: 1
 </td></tr>
<tr><td>4.</td><td>Create an Entity Link for the RightFax Server
An Entity Link establishes the details of how Entities will communicate with each other. Use the Add button to create a new link. In this case, Session Manager at the Main Site, <i>demoSM,</i> was configured to communicate with <i>RB_FaxServer</i> using <i>UDP</i> protocol over port <i>5060</i> as a Trusted Entity.


Entity Links

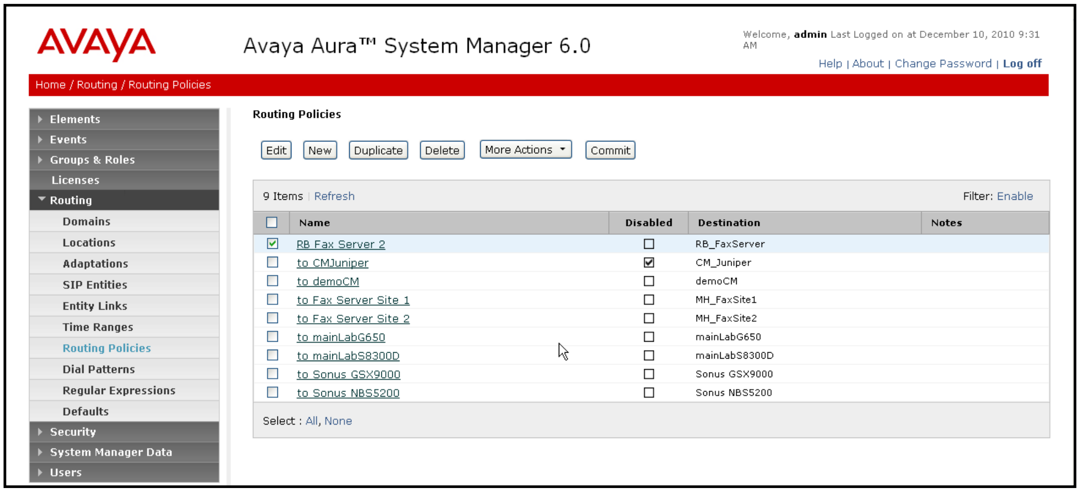
1 Item | Refresh Filter: Enable
<table>
<tr><th>☐</th><th>SIP Entity 1</th><th>Protocol</th><th>Port</th><th>SIP Entity 2</th><th>Port</th><th>Trusted</th></tr>
<tr><td>☐</td><td>demoSM</td><td>UDP</td><td>* 5060</td><td>RB_FaxServer</td><td>* 5060</td><td>☑</td></tr>
</table>
Select : All, None
* Input Required
 </td></tr>
</table>

Step **Description**

5.

Create a Routing Policy

Navigate to **Routing\Routing Policies** and click **New** to create a routing policy for incoming calls to the RightFax server. The illustration below was captured after the Policy ***RB_Fax_Server_2*** had been created and the following steps will describe how this policy was created.



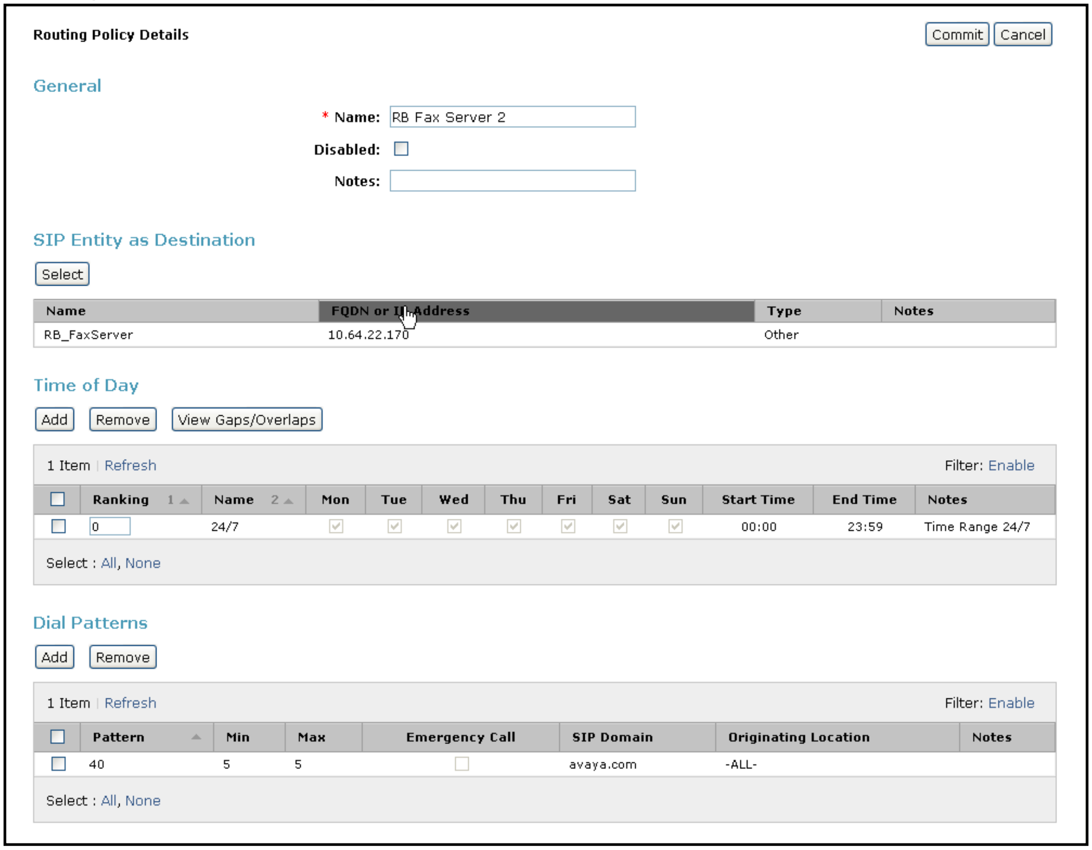
Routing Policies

9 Items | Refresh Filter: Enable

☐	Name	Disabled	Destination	Notes
☑	RB Fax Server 2	☐	RB_FaxServer	
☐	to CMJuniper	☑	CM_Juniper	
☐	to demoCM	☐	demoCM	
☐	to Fax Server Site 1	☐	MH_FaxSite1	
☐	to Fax Server Site 2	☐	MH_FaxSite2	
☐	to mainLabG650	☐	mainLabG650	
☐	to mainLabS8300D	☐	mainLabS8300D	
☐	to Sonus GSX9000	☐	Sonus GSX9000	
☐	to Sonus NBS5200	☐	Sonus NBS5200	

Select : All, None

A Routing Policy consists of a definition of the **SIP Entity as Destination**, the **Time of Day** the policy applies, and the **Dial Patterns** that will trigger this particular policy. Below are the settings used for this test. Use the **Select** or **Add** buttons to create or use existing definitions for each parameter.



Routing Policy Details

General

* Name: RB Fax Server 2

Disabled: ☐

Notes:

SIP Entity as Destination

Name	FQDN or IP Address	Type	Notes
RB_FaxServer	10.64.22.170	Other	

Time of Day

1 Item | Refresh Filter: Enable

☐	Ranking 1	Name 2	Mon	Tue	Wed	Thu	Fri	Sat	Sun	Start Time	End Time	Notes
☐	0	24/7	☑	☑	☑	☑	☑	☑	☑	00:00	23:59	Time Range 24/7

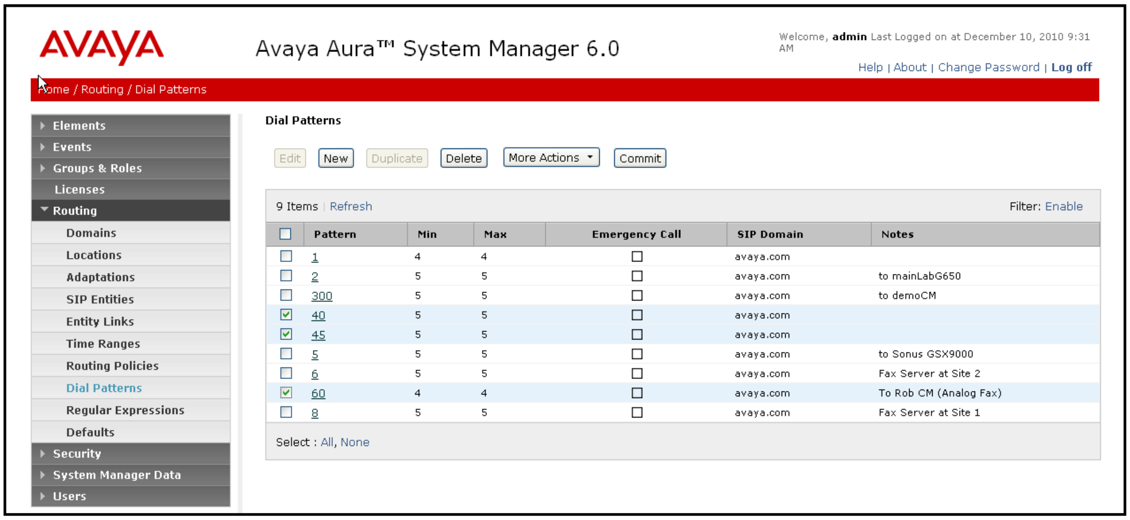
Select : All, None

Dial Patterns

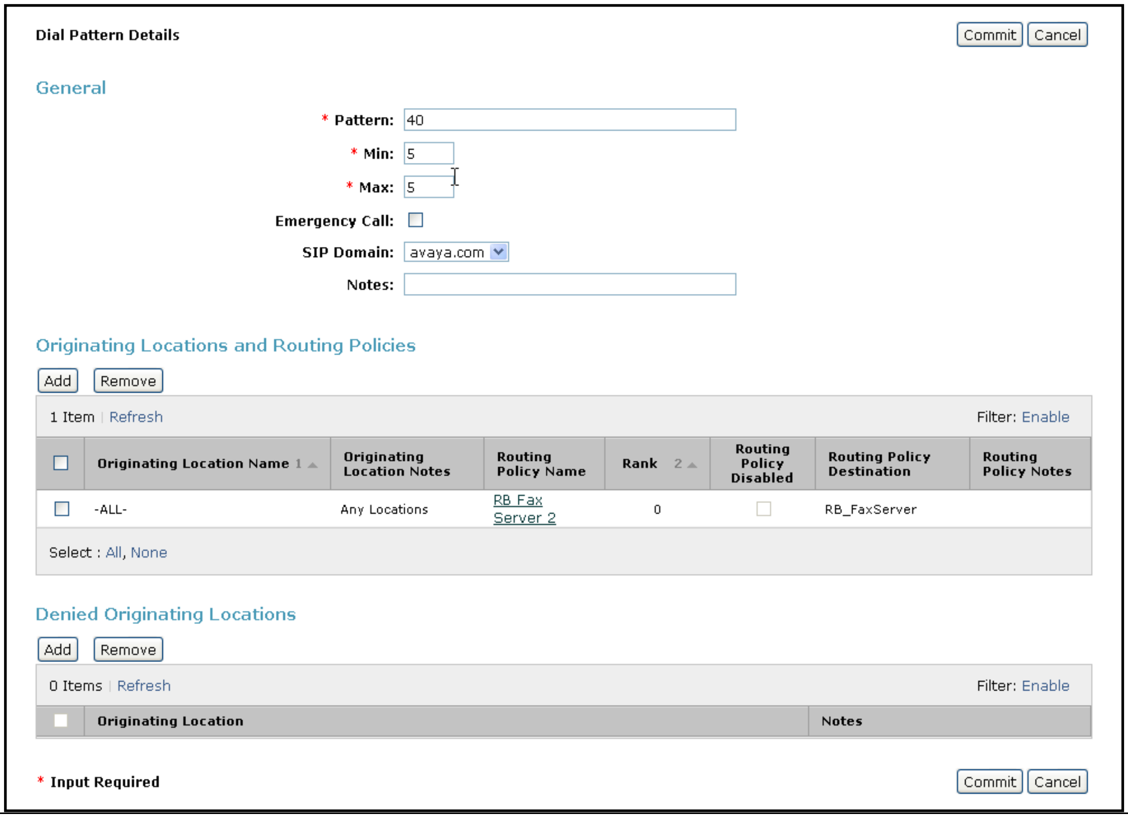
1 Item | Refresh Filter: Enable

☐	Pattern	Min	Max	Emergency Call	SIP Domain	Originating Location	Notes
☐	40	5	5	☐	avaya.com	-ALL-	

Select : All, None

Step	Description
6.	**Create or Modify Dial Patterns** Associating a dial pattern with a SIP Entity Link instructs Session Manager how to route calls matching the administered Dial Pattern(s). In the test, existing Routing Policies were modified for routing to endpoints or Entities at the Remote Site, and new Dial Patterns were created to route to the Main Site and Remote Site Fax Servers using the existing and new routing policies. In the snapshot below, the Dial Patterns were previously created. The applicable patterns were all 5 Digit extension patterns; dialed numbers beginning with 2 (the local analog fax machine at the Main Site), dialed numbers beginning with 40 (to route incoming Fax calls to the RightFax server at the Main Site), dialed numbers beginning with 45 (to route to Communication Manager at the Main Site in order to route via the public network interface between the sites). In addition, an existing 4 digit patterns beginning with 60 was used to route Fax calls to Communication Manager at the Main Site for routing via the public network interfaces to the analog machine at the Remote Site. The **‘40’ and ‘45’** dial patterns were created for this test, all others were in place in the test environment. 

☐	Pattern	Min	Max	Emergency Call	SIP Domain	Notes
☐	1	4	4	☐	avaya.com	
☐	2	5	5	☐	avaya.com	to mainLabG650
☐	300	5	5	☐	avaya.com	to demoCM
☑	40	5	5	☐	avaya.com	
☑	45	5	5	☐	avaya.com	
☐	5	5	5	☐	avaya.com	to Sonus GSX9000
☐	6	5	5	☐	avaya.com	Fax Server at Site 2
☑	60	4	4	☐	avaya.com	To Rob CM (Analog Fax)
☐	8	5	5	☐	avaya.com	Fax Server at Site 1

Step	Description
7.	**Create or Modify Dial Patterns – Continued** The entries required to create the new Dial Pattern for routing calls to the RightFax server at the Main site are illustrated below. The **Pattern, Min** and **Max** number of digits, and **SIP Domain** entries were used for this Dial Pattern definition. Click **Add** to associate the dial pattern with an existing Routing Policy, in this case the ***RB_Fax_Server_2*** policy created in **Step 5** above. **The Originating Location Name *All*** was used in this case to apply this pattern regardless of originating locations. In the same way, a new Dial Pattern was created and associated with the existing policy to route calls to Communication Manager at the Main Site using the Dial Pattern 45. This was used to route calls from the Main Site Fax Server to the Remote Site Fax Server. 

Dial Pattern Details

General

* **Pattern:** 40
* **Min:** 5
* **Max:** 5
Emergency Call: ☐
SIP Domain: avaya.com
Notes:

Originating Locations and Routing Policies

1 Item | Refresh   Filter: Enable

☐	Originating Location Name 1	Originating Location Notes	Routing Policy Name	Rank 2	Routing Policy Disabled	Routing Policy Destination	Routing Policy Notes
☐	-ALL-	Any Locations	RB Fax Server 2	0	☐	RB_FaxServer	

Select : All, None

Denied Originating Locations

0 Items | Refresh   Filter: Enable

☐	Originating Location	Notes

* **Input Required**

6. Configure Open Text Fax Server, RightFax Edition

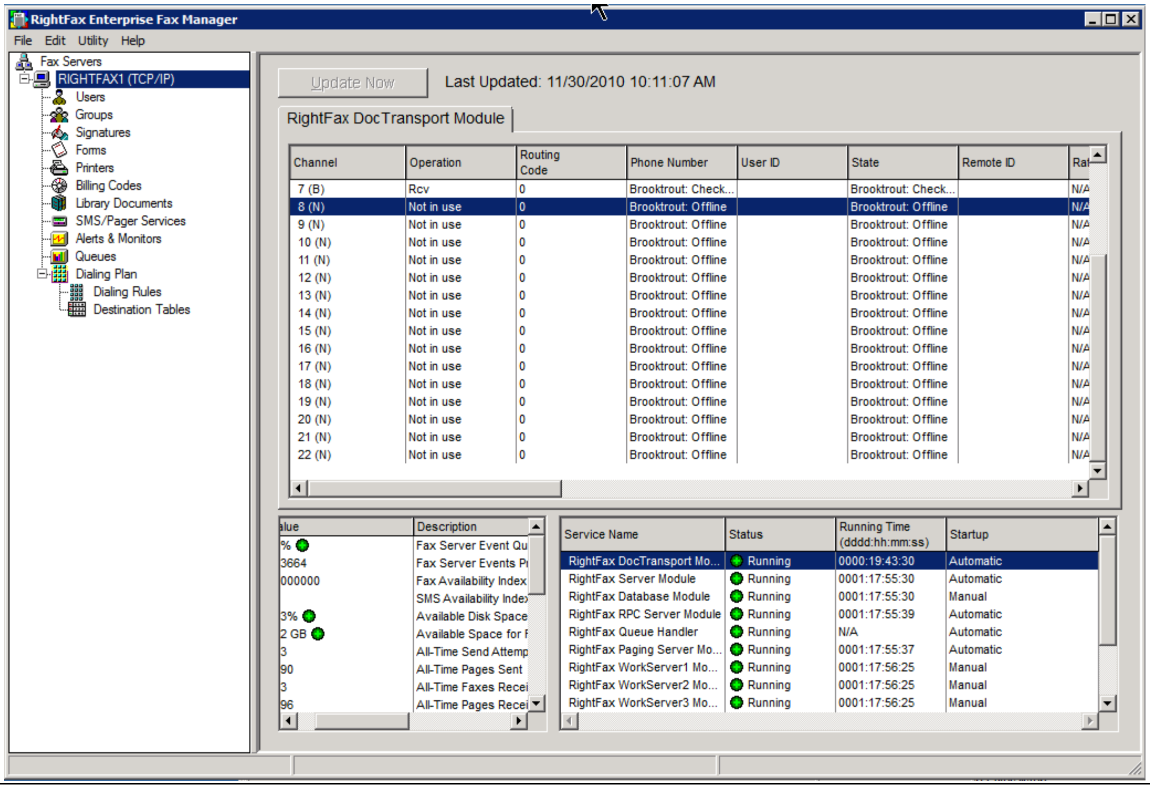
This section describes the configuration of Open Text Fax Server, RightFax Edition and the embedded Brooktrout SR140 virtual fax board software from Dialogic. It assumes that the application and all required software components, including Brooktrout SR140 and the database software (MSSQL 2008), have been installed and properly licensed. For instructions on installing RightFax, consult the RightFax Installation Guide [5].

Note that the configurations documented in this section pertain to interoperability between RightFax and the Avaya SIP infrastructure. The standard configurations pertaining to RightFax itself (e.g., administering fax channels) are not covered. For instructions on administering and operating RightFax, consult the RightFax Administrator's Guide [4].

The configuration procedures covered in this section include the following:

- Launch RightFax Enterprise Fax Manager and SR140 Configuration Tool (Steps 1 – 4)
- Configure SR140 IP stack (Step 5)
- Configure SR140 SIP IP parameters (Step 6)
- Configure SR140 T.38 parameters (Step 7)
- Update SR140 configuration file (Step 8)
- Administer RightFax dialing rules (Steps 9 – 11)
- Administer RightFax users (Steps 12 – 15)

The examples shown in this section refer to the Main Site. Unless specified otherwise, these same steps also apply to Remote Site using values appropriate from **Figure 1**.

Step	Description
1.	**Launch RightFax Enterprise Fax Manager** The RightFax configuration is performed using the RightFax Enterprise Fax Manager. Launch the RightFax Enterprise Fax Manager from the Windows Start menu. At the main window, highlight the host name of the fax server (created during the installation process) from the navigation menu in the left pane: 

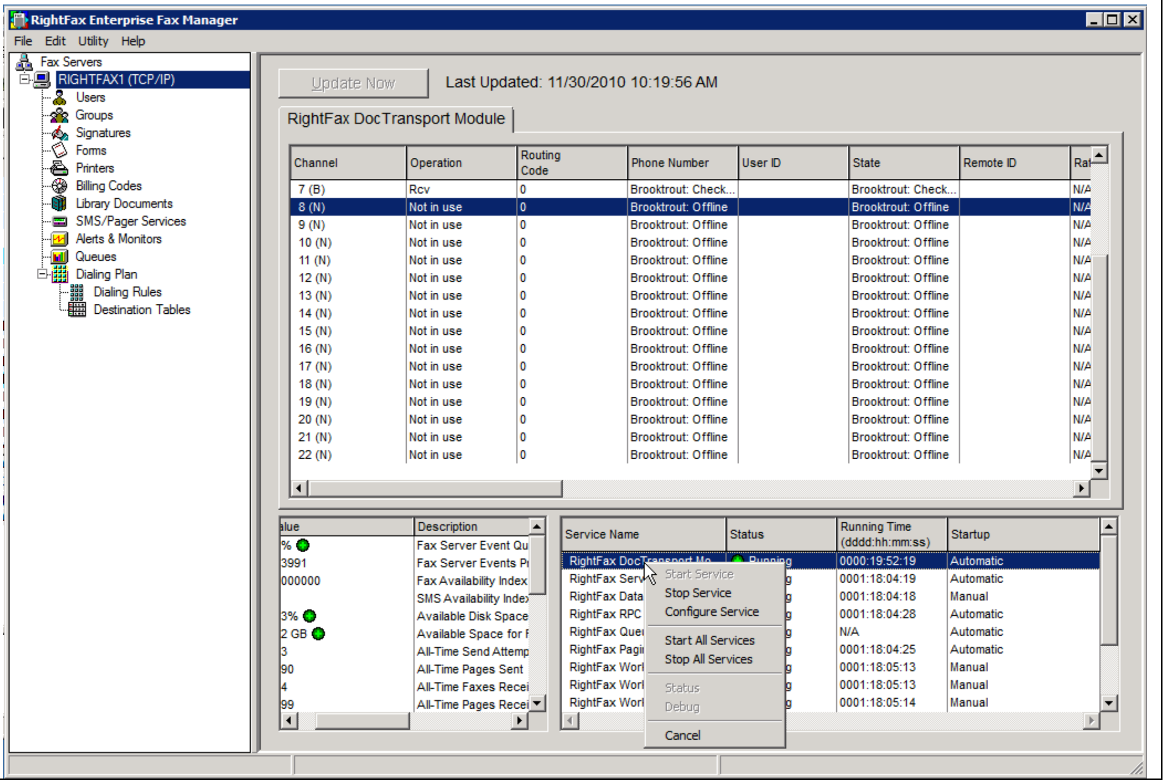
Last Updated: 11/30/2010 10:11:07 AM

RightFax DocTransport Module

Channel	Operation	Routing Code	Phone Number	User ID	State	Remote ID	Ra
7 (B)	Rcv	0	Brooktrout: Check...		Brooktrout: Check...		N/A
8 (N)	Not in use	0	Brooktrout: Offline		Brooktrout: Offline		N/A
9 (N)	Not in use	0	Brooktrout: Offline		Brooktrout: Offline		N/A
10 (N)	Not in use	0	Brooktrout: Offline		Brooktrout: Offline		N/A
11 (N)	Not in use	0	Brooktrout: Offline		Brooktrout: Offline		N/A
12 (N)	Not in use	0	Brooktrout: Offline		Brooktrout: Offline		N/A
13 (N)	Not in use	0	Brooktrout: Offline		Brooktrout: Offline		N/A
14 (N)	Not in use	0	Brooktrout: Offline		Brooktrout: Offline		N/A
15 (N)	Not in use	0	Brooktrout: Offline		Brooktrout: Offline		N/A
16 (N)	Not in use	0	Brooktrout: Offline		Brooktrout: Offline		N/A
17 (N)	Not in use	0	Brooktrout: Offline		Brooktrout: Offline		N/A
18 (N)	Not in use	0	Brooktrout: Offline		Brooktrout: Offline		N/A
19 (N)	Not in use	0	Brooktrout: Offline		Brooktrout: Offline		N/A
20 (N)	Not in use	0	Brooktrout: Offline		Brooktrout: Offline		N/A
21 (N)	Not in use	0	Brooktrout: Offline		Brooktrout: Offline		N/A
22 (N)	Not in use	0	Brooktrout: Offline		Brooktrout: Offline		N/A

alue	Description
% ●	Fax Server Event Qu
3664	Fax Server Events Pr
000000	Fax Availability Index
	SMS Availability Index
3% ●	Available Disk Space
2 GB ●	Available Space for F
3	All-Time Send Attemp
90	All-Time Pages Sent
3	All-Time Faxes Recei
96	All-Time Pages Recei

Service Name	Status	Running Time (dddd:hh:mm:ss)	Startup
RightFax DocTransport Mo...	● Running	0000:19:43:30	Automatic
RightFax Server Module	● Running	0001:17:55:30	Automatic
RightFax Database Module	● Running	0001:17:55:30	Manual
RightFax RPC Server Module	● Running	0001:17:55:39	Automatic
RightFax Queue Handler	● Running	N/A	Automatic
RightFax Paging Server Mo...	● Running	0001:17:55:37	Automatic
RightFax WorkServer1 Mo...	● Running	0001:17:56:25	Manual
RightFax WorkServer2 Mo...	● Running	0001:17:56:25	Manual
RightFax WorkServer3 Mo...	● Running	0001:17:56:25	Manual

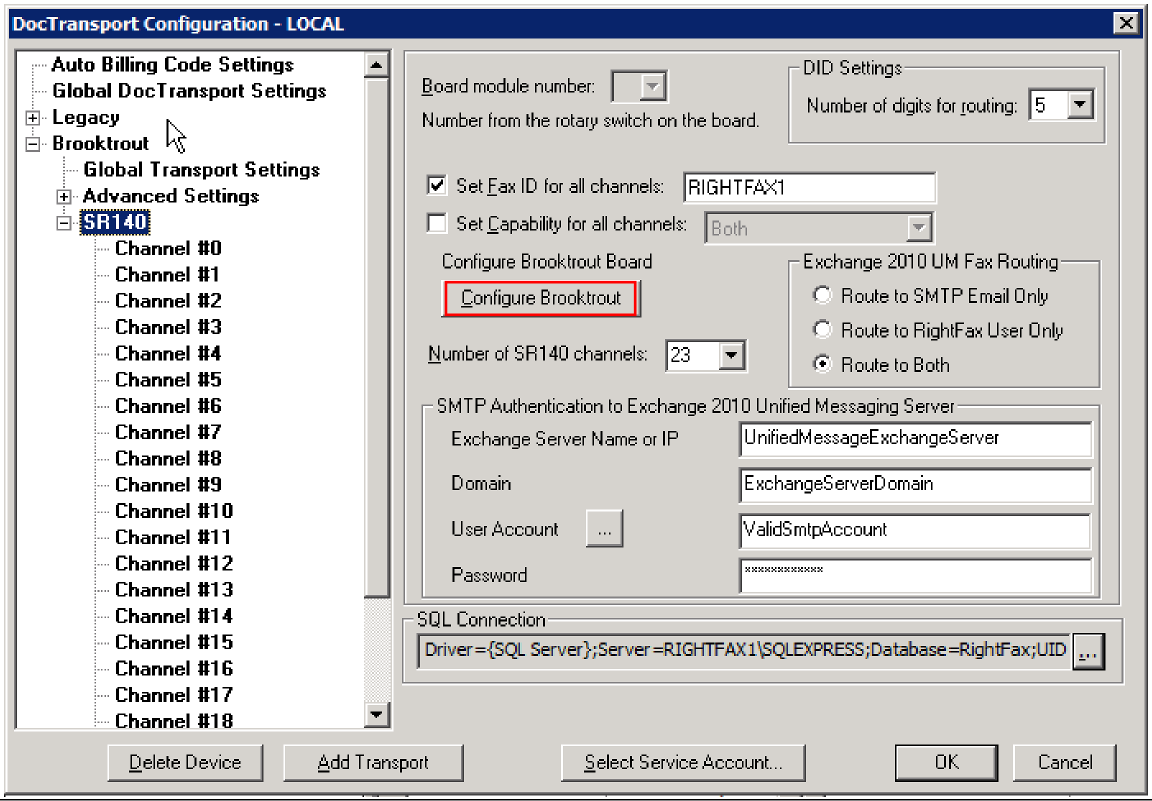
Step	Description
2.	**RightFax DocTransport Module** The Brooktrout SR140 was configured during installation. To view or modify the settings, the RightFax DocTransport Module must be stopped. Right-click this module in the lower right pane and select **Stop All Services**. After all the service modules indicate the stopped status, right-click the **RightFax DocTransport Module** name again to select **Configure Service**. 

Last Updated: 11/30/2010 10:19:56 AM

Channel	Operation	Routing Code	Phone Number	User ID	State	Remote ID	Ra
7 (B)	Rcv	0	Brooktrout: Check...		Brooktrout: Check...		N/A
8 (N)	Not in use	0	Brooktrout: Offline		Brooktrout: Offline		N/A
9 (N)	Not in use	0	Brooktrout: Offline		Brooktrout: Offline		N/A
10 (N)	Not in use	0	Brooktrout: Offline		Brooktrout: Offline		N/A
11 (N)	Not in use	0	Brooktrout: Offline		Brooktrout: Offline		N/A
12 (N)	Not in use	0	Brooktrout: Offline		Brooktrout: Offline		N/A
13 (N)	Not in use	0	Brooktrout: Offline		Brooktrout: Offline		N/A
14 (N)	Not in use	0	Brooktrout: Offline		Brooktrout: Offline		N/A
15 (N)	Not in use	0	Brooktrout: Offline		Brooktrout: Offline		N/A
16 (N)	Not in use	0	Brooktrout: Offline		Brooktrout: Offline		N/A
17 (N)	Not in use	0	Brooktrout: Offline		Brooktrout: Offline		N/A
18 (N)	Not in use	0	Brooktrout: Offline		Brooktrout: Offline		N/A
19 (N)	Not in use	0	Brooktrout: Offline		Brooktrout: Offline		N/A
20 (N)	Not in use	0	Brooktrout: Offline		Brooktrout: Offline		N/A
21 (N)	Not in use	0	Brooktrout: Offline		Brooktrout: Offline		N/A
22 (N)	Not in use	0	Brooktrout: Offline		Brooktrout: Offline		N/A

alue	Description
% ●	Fax Server Event Qu
3991	Fax Server Events Pr
000000	Fax Availability Index
	SMS Availability Index
3% ●	Available Disk Space
2 GB ●	Available Space for F
3	All-Time Send Attemp
90	All-Time Pages Sent
4	All-Time Faxes Recei
99	All-Time Pages Recei

Service Name	Status	Running Time (dddd:hh:mm:ss)	Startup
RightFax DocTransport Mo	● Running	0000:19:52:19	Automatic
RightFax Serv	g	0001:18:04:19	Automatic
RightFax Data	g	0001:18:04:18	Manual
RightFax RPC	g	0001:18:04:28	Automatic
RightFax Quer	g	N/A	Automatic
RightFax Pagi	g	0001:18:04:25	Automatic
RightFax Worl	g	0001:18:05:13	Manual
RightFax Worl	g	0001:18:05:13	Manual
RightFax Worl	g	0001:18:05:14	Manual

Step	Description
3.	**RightFax DocTransport Module - Continued** In the **DocTransport Configuration** window that appears, click the **Configure Brooktrout** button. 

Board module number:
Number from the rotary switch on the board.

DID Settings
Number of digits for routing: 5

☑ Set Fax ID for all channels: RIGHTFAX1
☐ Set Capability for all channels: Both

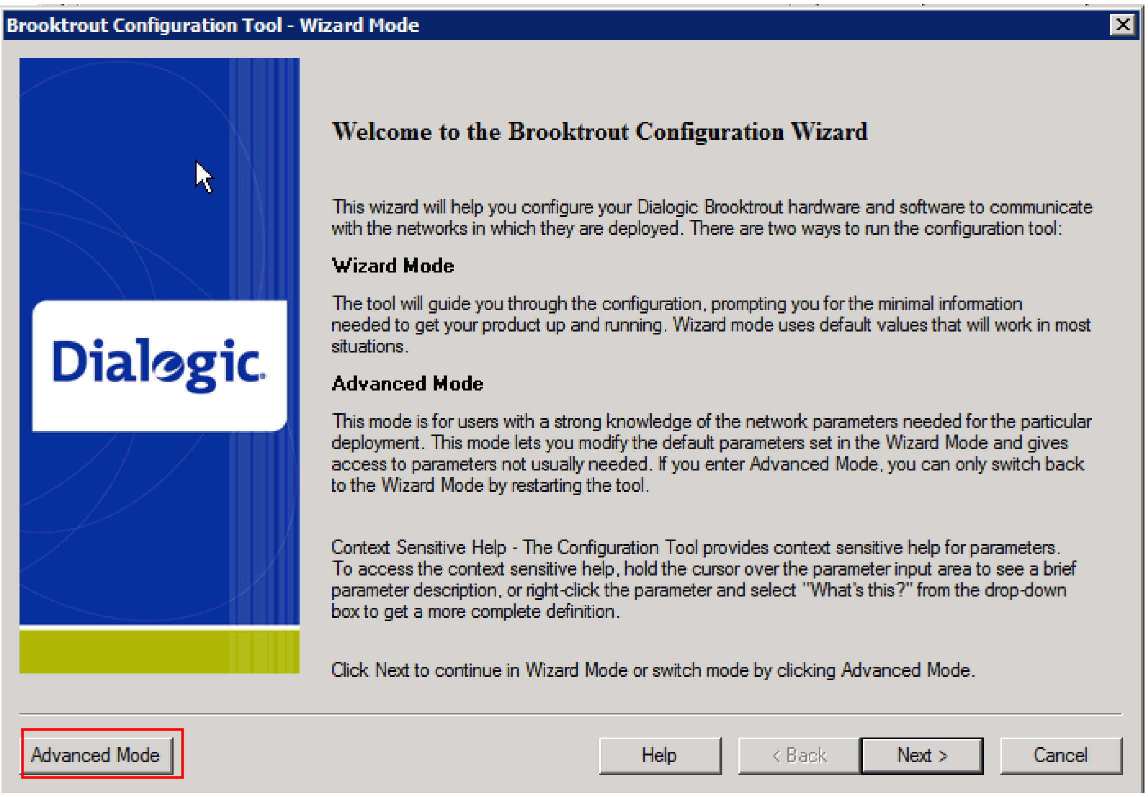
Configure Brooktrout Board

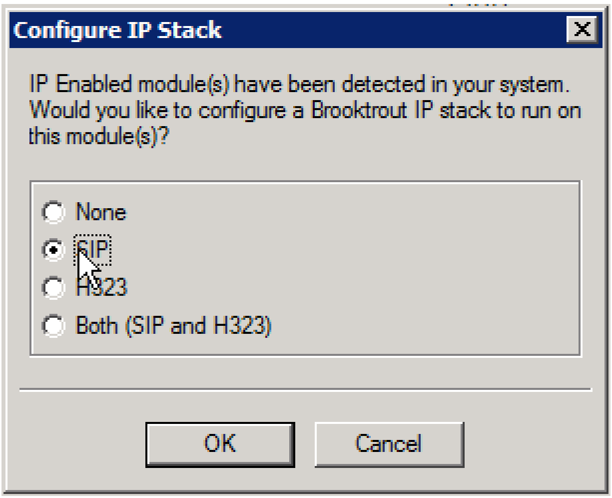
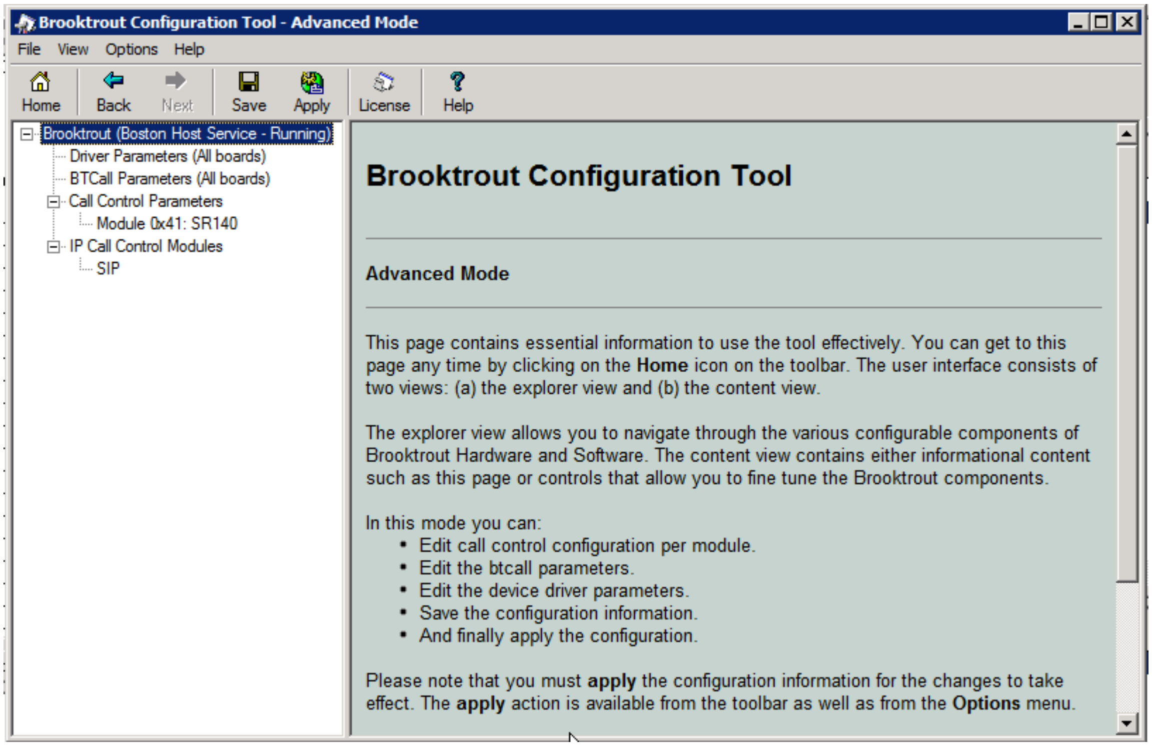
Number of SR140 channels: 23

Exchange 2010 UM Fax Routing
○ Route to SMTP Email Only
○ Route to RightFax User Only
● Route to Both

SMTP Authentication to Exchange 2010 Unified Messaging Server
Exchange Server Name or IP: UnifiedMessageExchangeServer
Domain: ExchangeServerDomain
User Account: ValidSmtpAccount
Password: ************

SQL Connection
Driver={SQL Server};Server=RIGHTFAX1\SQLEXPRESS;Database=RightFax;UID

Step	Description
4.	**Brooktrout Configuration Tool** The **Brooktrout Configuration Tool – Wizard Mode** window gets displayed. Click the **Advanced Mode** button in this window.  Brooktrout Configuration Tool - Wizard Mode Welcome to the Brooktrout Configuration Wizard This wizard will help you configure your Dialogic Brooktrout hardware and software to communicate with the networks in which they are deployed. There are two ways to run the configuration tool: **Wizard Mode** The tool will guide you through the configuration, prompting you for the minimal information needed to get your product up and running. Wizard mode uses default values that will work in most situations. **Advanced Mode** This mode is for users with a strong knowledge of the network parameters needed for the particular deployment. This mode lets you modify the default parameters set in the Wizard Mode and gives access to parameters not usually needed. If you enter Advanced Mode, you can only switch back to the Wizard Mode by restarting the tool. Context Sensitive Help - The Configuration Tool provides context sensitive help for parameters. To access the context sensitive help, hold the cursor over the parameter input area to see a brief parameter description, or right-click the parameter and select "What's this?" from the drop-down box to get a more complete definition. Click Next to continue in Wizard Mode or switch mode by clicking Advanced Mode.

Step	Description
5.	**Configure IP Stack** A Configure IP Stack window is displayed on first invocation of the Brooktrout SR140 configuration tool (assuming the Brooktrout SR-140 licenses were installed):  IP Enabled module(s) have been detected in your system. Would you like to configure a Brooktrout IP stack to run on this module(s)? ○ None ● SIP ○ H323 ○ Both (SIP and H323) Choose **SIP** and click **OK**. The following SR140 configuration tool window is displayed.  **Brooktrout Configuration Tool** **Advanced Mode** This page contains essential information to use the tool effectively. You can get to this page any time by clicking on the **Home** icon on the toolbar. The user interface consists of two views: (a) the explorer view and (b) the content view. The explorer view allows you to navigate through the various configurable components of Brooktrout Hardware and Software. The content view contains either informational content such as this page or controls that allow you to fine tune the Brooktrout components. In this mode you can: • Edit call control configuration per module. • Edit the btcall parameters. • Edit the device driver parameters. • Save the configuration information. • And finally apply the configuration. Please note that you must **apply** the configuration information for the changes to take effect. The **apply** action is available from the toolbar as well as from the **Options** menu. Note that IP Stack can be viewed/reconfigured from the Brooktrout Configuration Tool menu **Options → Configure IP Stack**.

Step	Description
6.	**Configure SIP IP Parameters**

Important*: This step describes configuring the SIP Primary Gateway address using the Brooktrout Configuration Tool. This method is sufficient if the RightFax Server will communicate with a single SIP gateway or Session Manager. Alternatively, use RightFax Dialing Rules, described in steps 10-11 below, to specify the address of the SIP Gateway. You should only use one of these options. Do not use both.*

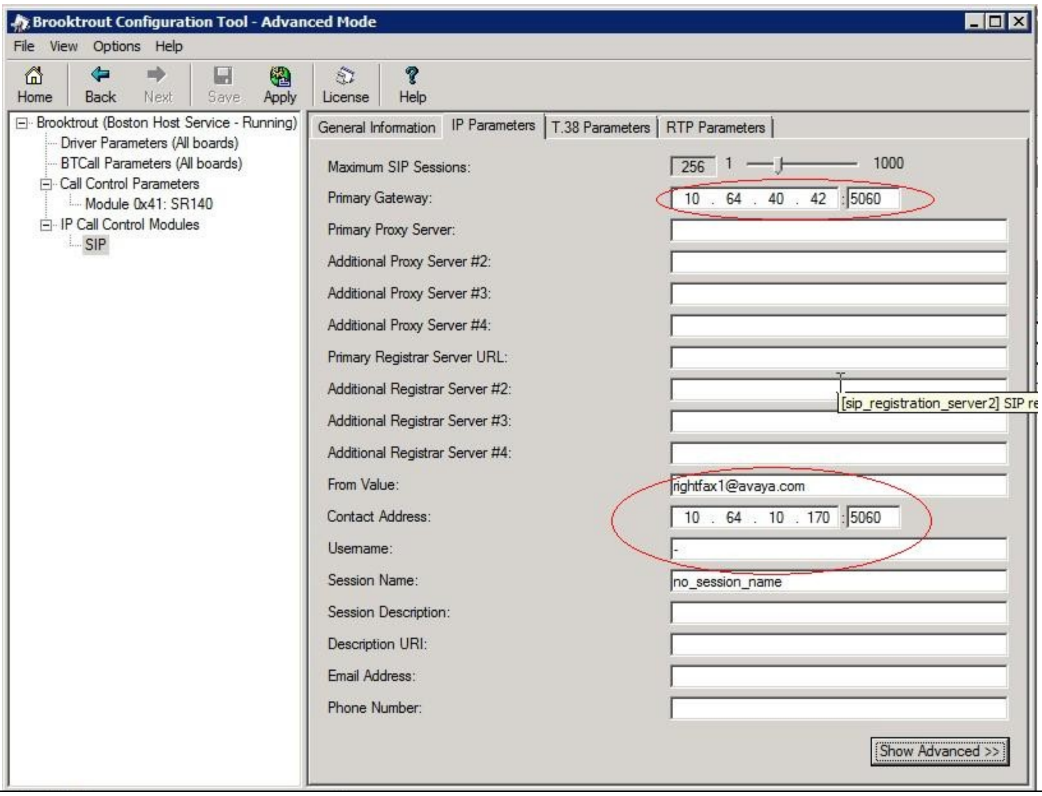
Either option listed above is appropriate if Media Shuffling is enabled in the Avaya communications equipment.

If the RightFax Server needs to be configured to communicate directly with multiple gateways or Session Managers, use only RightFax Dialing Rules. Use a separate Dialing Rule for gateway.

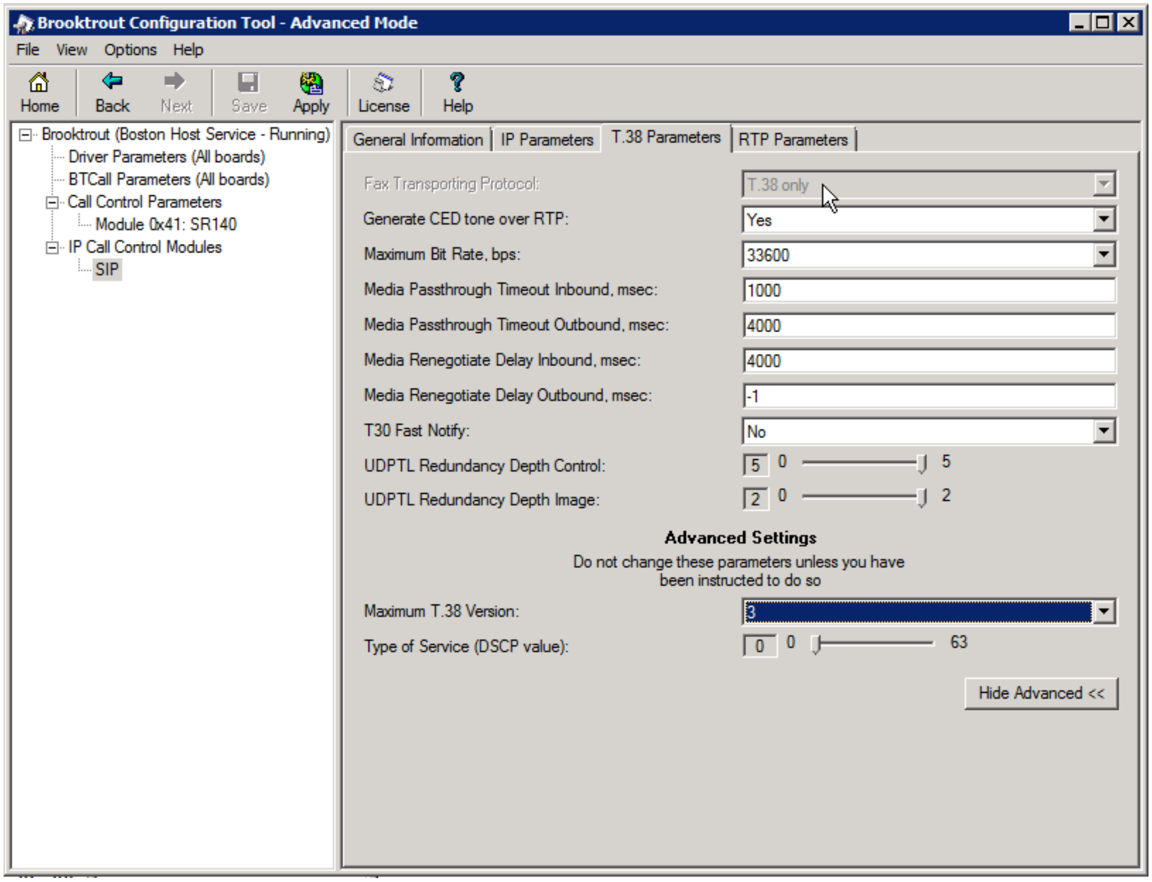
Navigate to **Brooktrout → IP Call Control Modules → SIP** in the left navigation menu. Select the **IP Parameters** tab in the right pane. Configure the fields as follows:

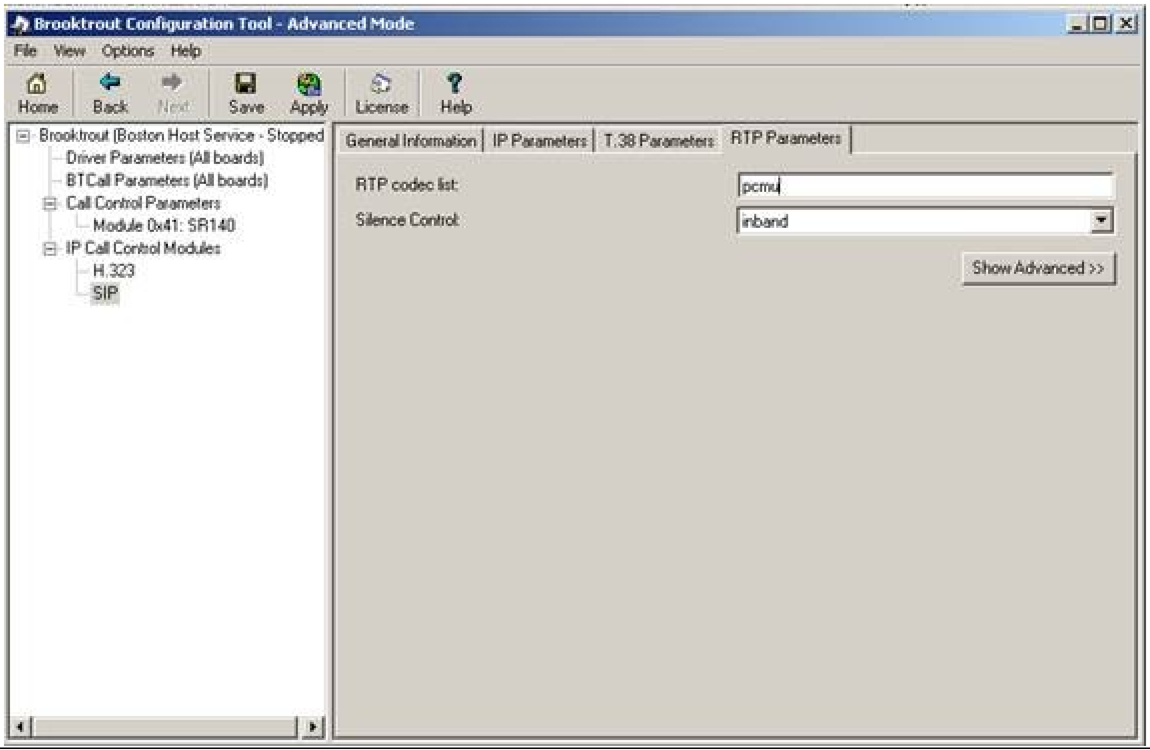
- **Primary Gateway** –set to the IP address of the Session Manager server, and port number ***5060***.
- **From Value** – set to appropriate ***UserInfo@DomainName.*** The *DomainName* should be set to the authoritative domain as configured in Session Manager.
- **Contact Address** – set to the IP address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

Use default values for all other fields.



Maximum SIP Sessions: 256 1 1000
Primary Gateway: 10 . 64 . 40 . 42 :5060
Primary Proxy Server:
Additional Proxy Server #2:
Additional Proxy Server #3:
Additional Proxy Server #4:
Primary Registrar Server URL:
Additional Registrar Server #2:
Additional Registrar Server #3:
Additional Registrar Server #4:
From Value: rightfax1@avaya.com
Contact Address: 10 . 64 . 10 . 170 :5060
Username: -
Session Name: no_session_name
Session Description:
Description URI:
Email Address:
Phone Number:

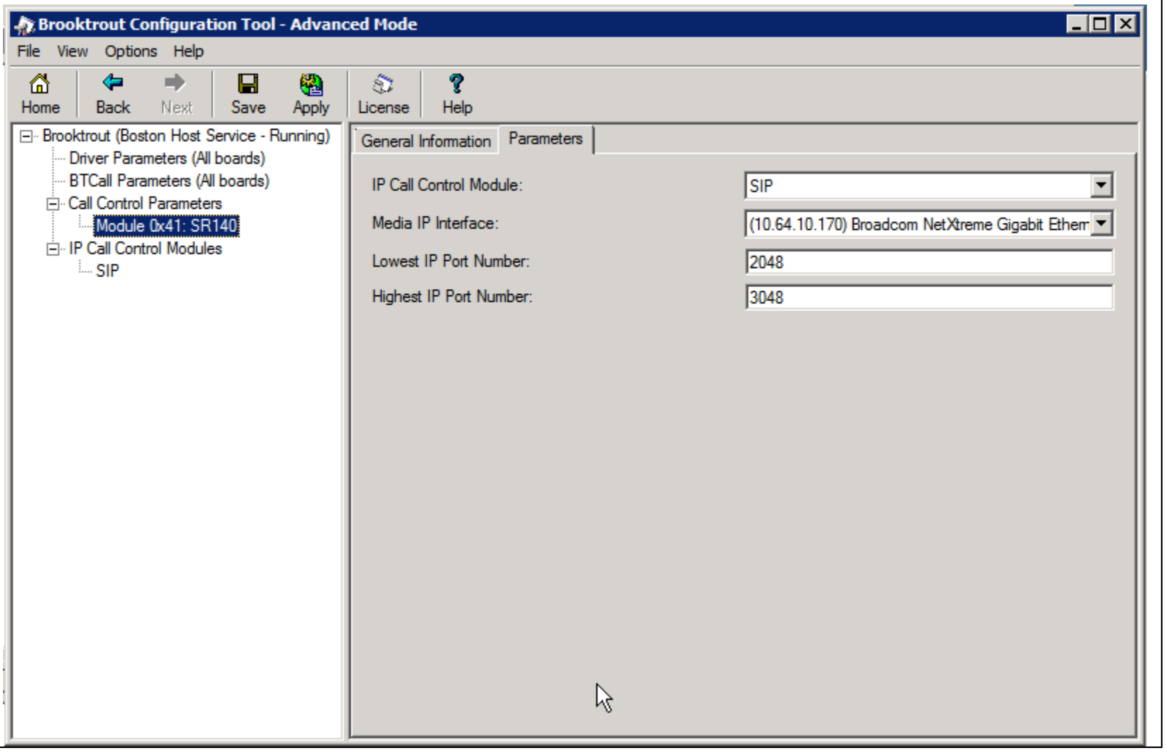
Step	Description
7.	**Configure T.38 Parameters** Select the **T.38 Parameters** tab. Configure the fields as shown below in the screenshot. *Notes:* - *"Media Renegotiate Delay Inbound, msec" has been changed from default.* - *"Maximum Bit Rate, bps" is set to maximum, 33600.* - *"Maximum T.38 Version" is set to "3".*  Fax Transporting Protocol: T.38 only Generate CED tone over RTP: Yes Maximum Bit Rate, bps: 33600 Media Passthrough Timeout Inbound, msec: 1000 Media Passthrough Timeout Outbound, msec: 4000 Media Renegotiate Delay Inbound, msec: 4000 Media Renegotiate Delay Outbound, msec: -1 T30 Fast Notify: No UDPTL Redundancy Depth Control: 5 0 5 UDPTL Redundancy Depth Image: 2 0 2 **Advanced Settings** Do not change these parameters unless you have been instructed to do so Maximum T.38 Version: 3 Type of Service (DSCP value): 0 0 63

Step	Description
8.	**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.  RTP codec list: pcmu Silence Control: inband

9. **Configure RTP Port Range**

In the left-hand navigation area, expand **Call Control Parameters** and click the **Module 0x##: SR140** item for your FoIP channels.

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: 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 set to 3048.*



IP Call Control Module: SIP

Media IP Interface: (10.64.10.170) Broadcom NetXtreme Gigabit Etherr

Lowest IP Port Number: 2048

Highest IP Port Number: 3048

10. **Complete Brooktrout SR140 Configuration**

After verifying all the above parameters are properly set, click **Save** in the button menu. Exit the Brooktrout Configuration Tool.

In the **DocTransport Configuration** screen, click the **OK** button.

From Windows explorer, navigate to the Brooktrout folder in the RightFax install directory (typically Program Files\RightFax\DecTransport\Brooktrout), Open and edit the **callctrl.cfg** file as follows, then save the updates:

- Verify that the following configuration segment is present; and that rtp_codec matches the value specified in **Step 8** above, either “pcmu” or “pcma”.

```
[host_module.1/rtp]
 rtp_codec=pcmu
```

- Change **rtp_ced_enable** setting to ***true*** under the **[host_modele.1/t.38parameters]** header (. . . below indicates other entries under the header)

```
[host_module.1/t.38parameters]
 ...
 rtp_ced_enable=true
 ...
```

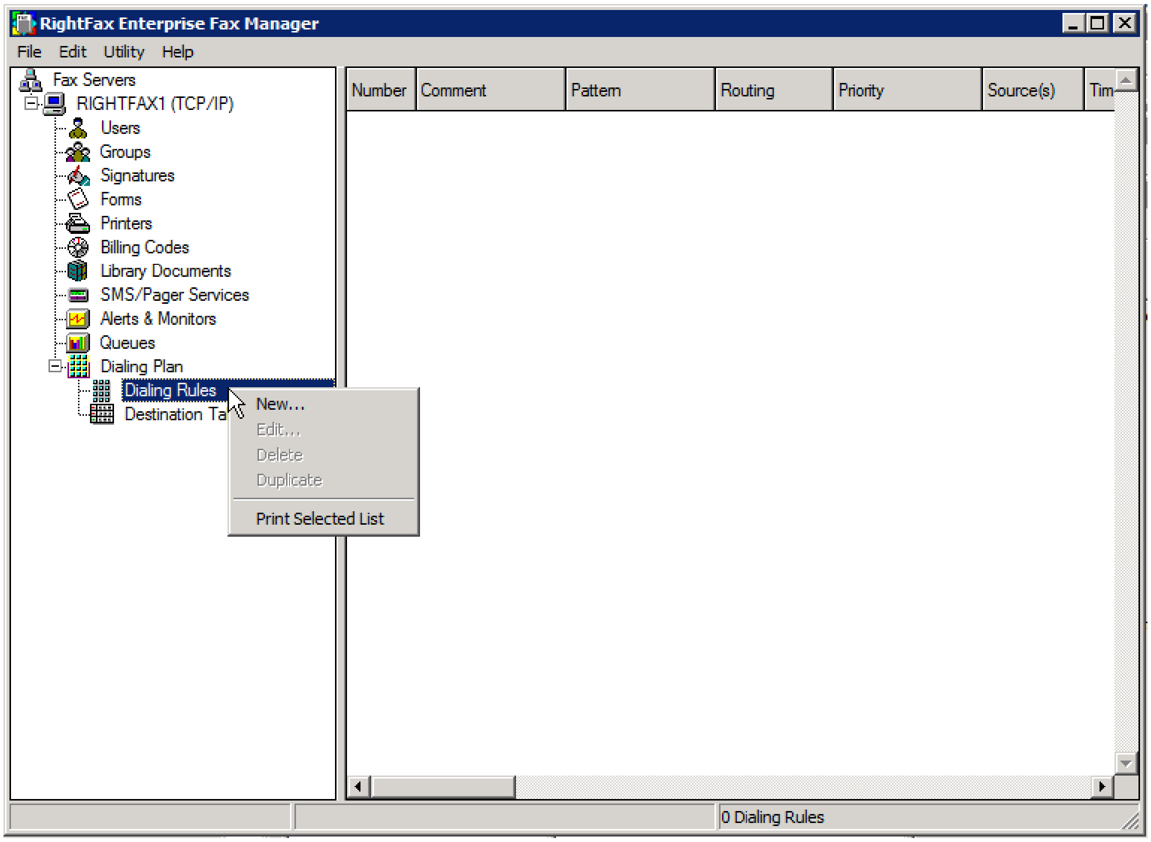
After making and saving the above updates in the **callctrl.cfg** file, restart all RightFax service modules by right clicking the **RightFax DocTransport Module** name in the lower right pane of the RightFax Enterprise Fax Manager window and select **Start All Services** (see **Step 2**).

11. **Configure Dialing Rules**

Note: See Step 6 above for discussion on using RightFax Dialing Rules versus the Brooktrout Configuration Tool to configure media gateway addresses.

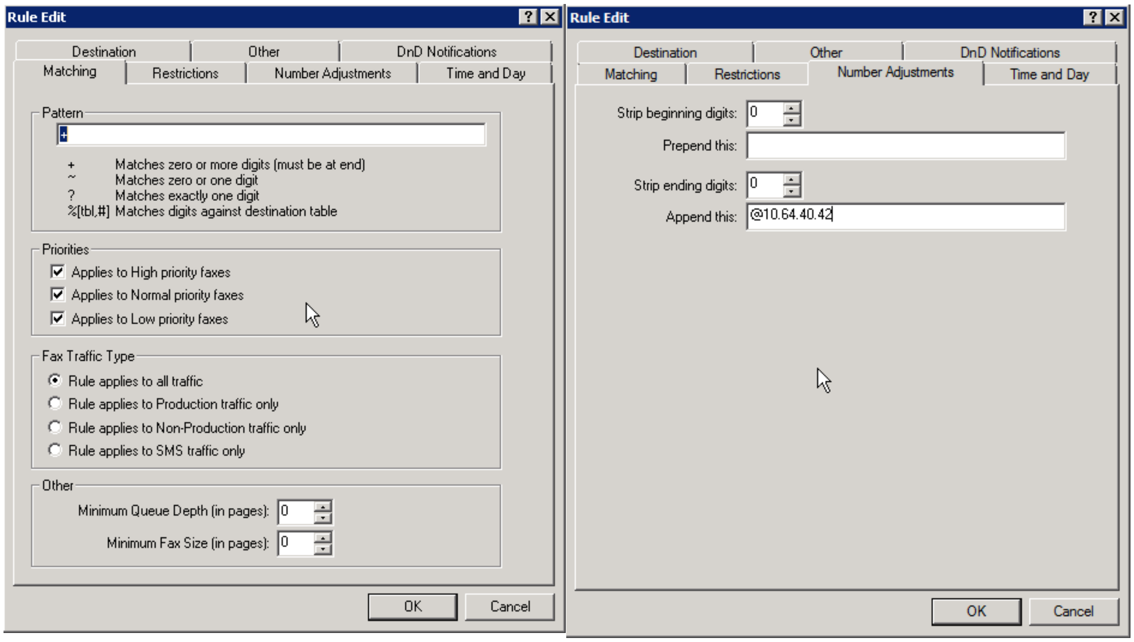
Dialing Rules are used by RightFax to route calls. In the compliance test, a dialing rule was created to route outbound fax calls to the Avaya Media Gateway. In the left navigation menu under the host name of the fax server, navigate to **Dialing Plan**, right-click **Dialing Rules** and select **New** to create a new rule.

The example below shows the single rule created for the compliance test at site 1. The * in the **Pattern** field indicates that this rule applies to all dialed numbers. To view the details, double click on the rule in the right pane.



12. **Configure Dialing Rules - Continued**

The **Rule Edit** window will appear as shown below. The **Number Adjustments** tab shows the digit string manipulation that is done to each dialed number. In the example below, each number is appended with ***@10.64.40.42*** as indicated in the **Append this** field. This IP address is for the Session Manager server at the Main Site.



Pattern: +

+	Matches zero or more digits (must be at end)
~	Matches zero or one digit
?	Matches exactly one digit
%[tbl,#]	Matches digits against destination table

Priorities
- ☑ Applies to High priority faxes
- ☑ Applies to Normal priority faxes
- ☑ Applies to Low priority faxes

Fax Traffic Type
- ● Rule applies to all traffic
- ○ Rule applies to Production traffic only
- ○ Rule applies to Non-Production traffic only
- ○ Rule applies to SMS traffic only

Other

Minimum Queue Depth (in pages): 0

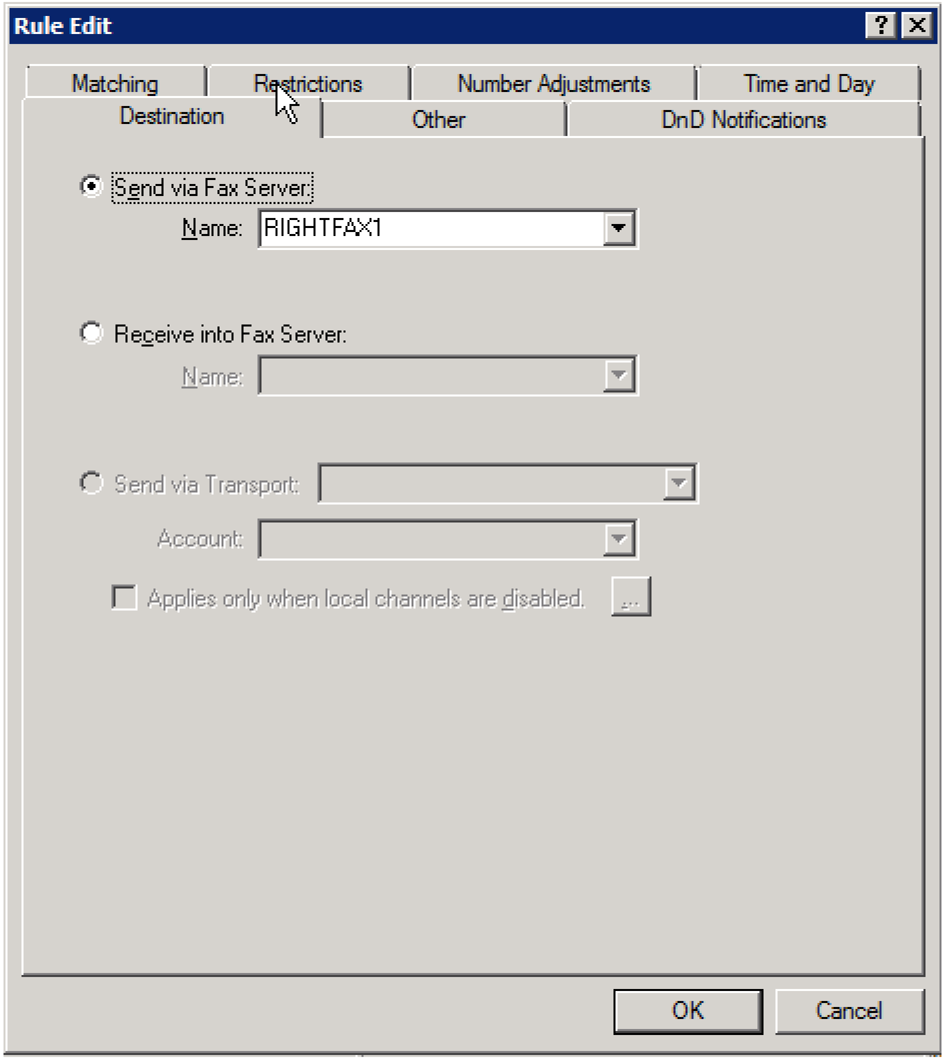
Minimum Fax Size (in pages): 0

Strip beginning digits: 0

Prepend this:

Strip ending digits: 0

Append this: @10.64.40.42

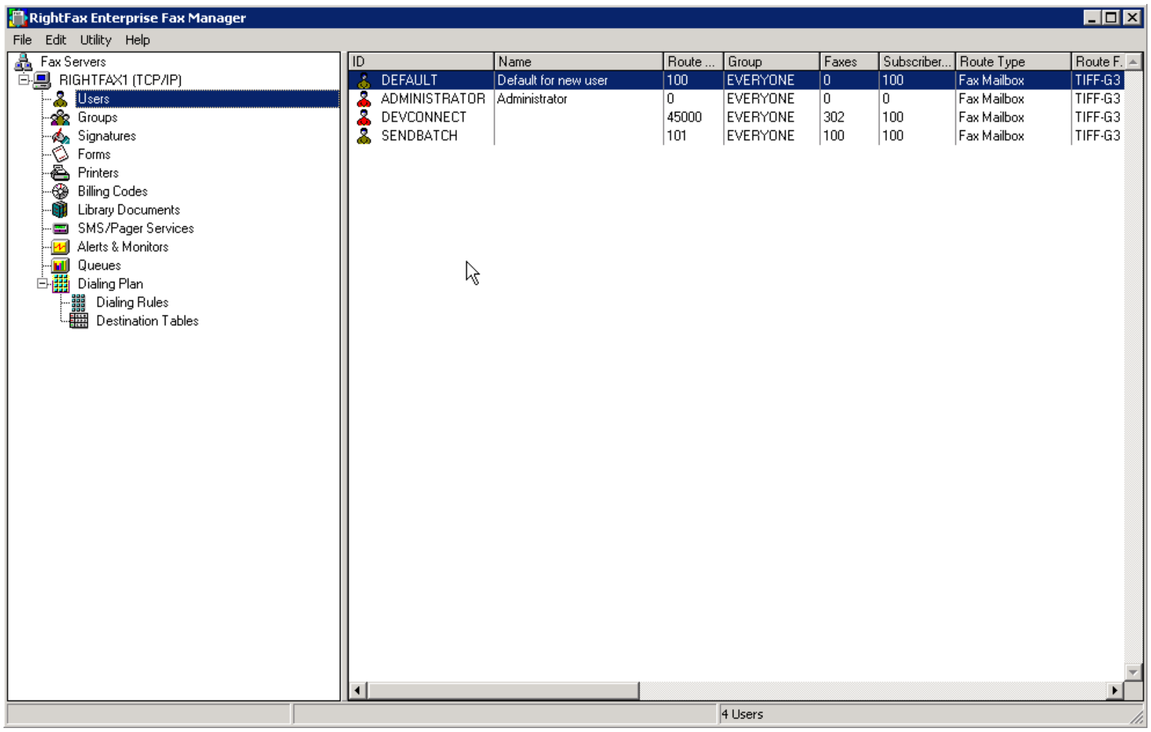
13.	**Configure Dialing Rules - Continued** In the **Destination** tab of the **Rule Edit** window, select the correct host name of the fax server for **Send via Fax Server**. This is typically the local computer name. 

● Send via Fax Server:

Name: RIGHTFAX1

14. **Configure Users**

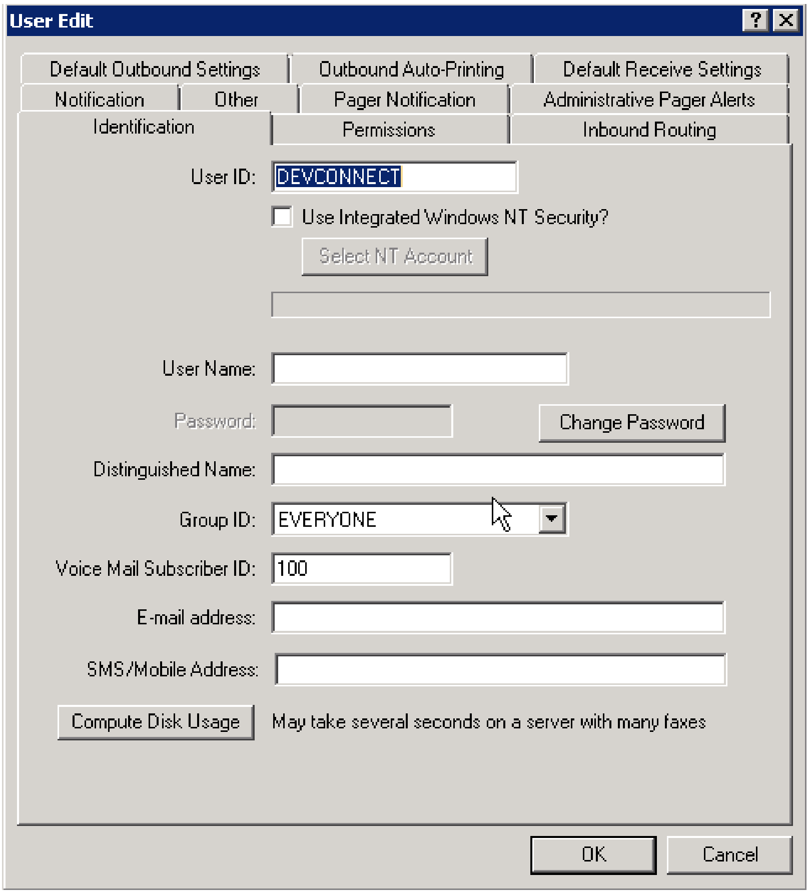
A user is created on RightFax for each incoming fax number. The user represents the fax recipient. To view the list of users, navigate to **Users** in the left navigation menu under the host name of the fax server. The example below shows a list of 4 users.To view the details of a user, double-click on the user entry in the right pane.



ID	Name	Route ...	Group	Faxes	Subscriber...	Route Type	Route F.
DEFAULT	Default for new user	100	EVERYONE	0	100	Fax Mailbox	TIFF-G3
ADMINISTRATOR	Administrator	0	EVERYONE	0	0	Fax Mailbox	TIFF-G3
DEVCONNECT		45000	EVERYONE	302	100	Fax Mailbox	TIFF-G3
SENDBATCH		101	EVERYONE	100	100	Fax Mailbox	TIFF-G3

15. **Configure Users – Identification**

The **User Edit** window will appear as shown below. Select the **Identification** tab. The example below shows the settings used for the compliance test at the Main Sites. The **User ID** field is set to a descriptive name. Appropriate values should be entered or selected for other fields.



User ID: DEVCONNECT

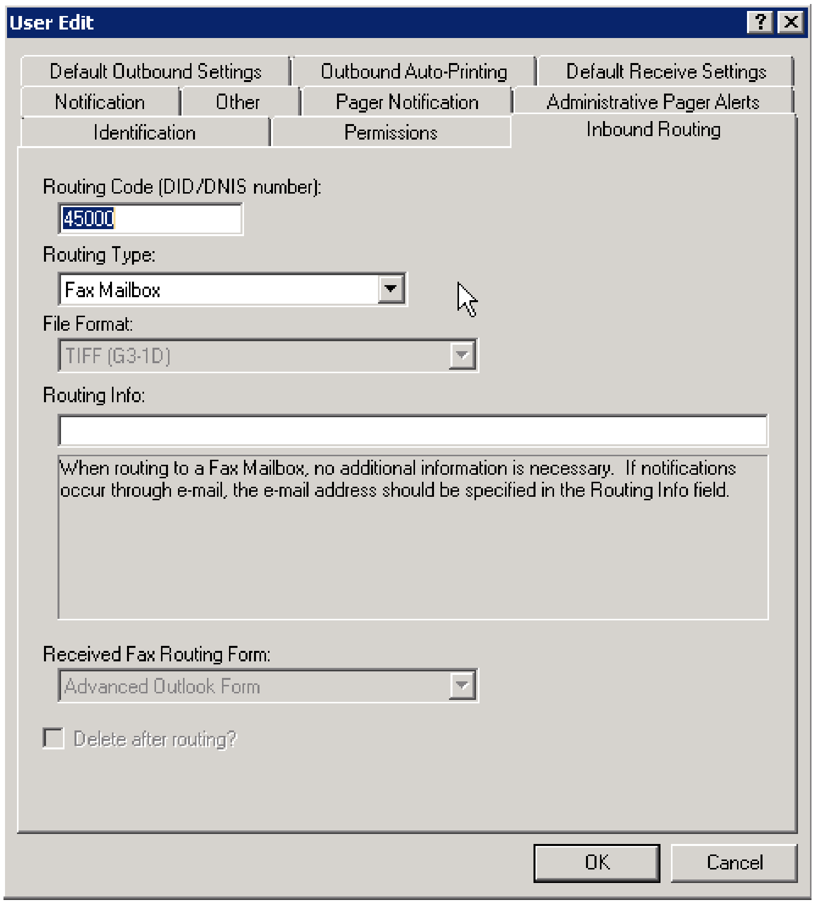
☐ Use Integrated Windows NT Security?

Group ID: EVERYONE

Voice Mail Subscriber ID: 100

16. **Configure Users – Inbound Routing**

On the **Inbound Routing** tab, the **Routing Code** field is set to the fax number of the recipient. In the case of the compliance test, this was extension ***40000*** for the Main Site, and ***45000*** for the Remote Site (pictured below). Default values may be used for all other fields.



Routing Code (DID/DNIS number): 45000

Routing Type: Fax Mailbox

File Format: TIFF (G3-1D)

Routing Info:

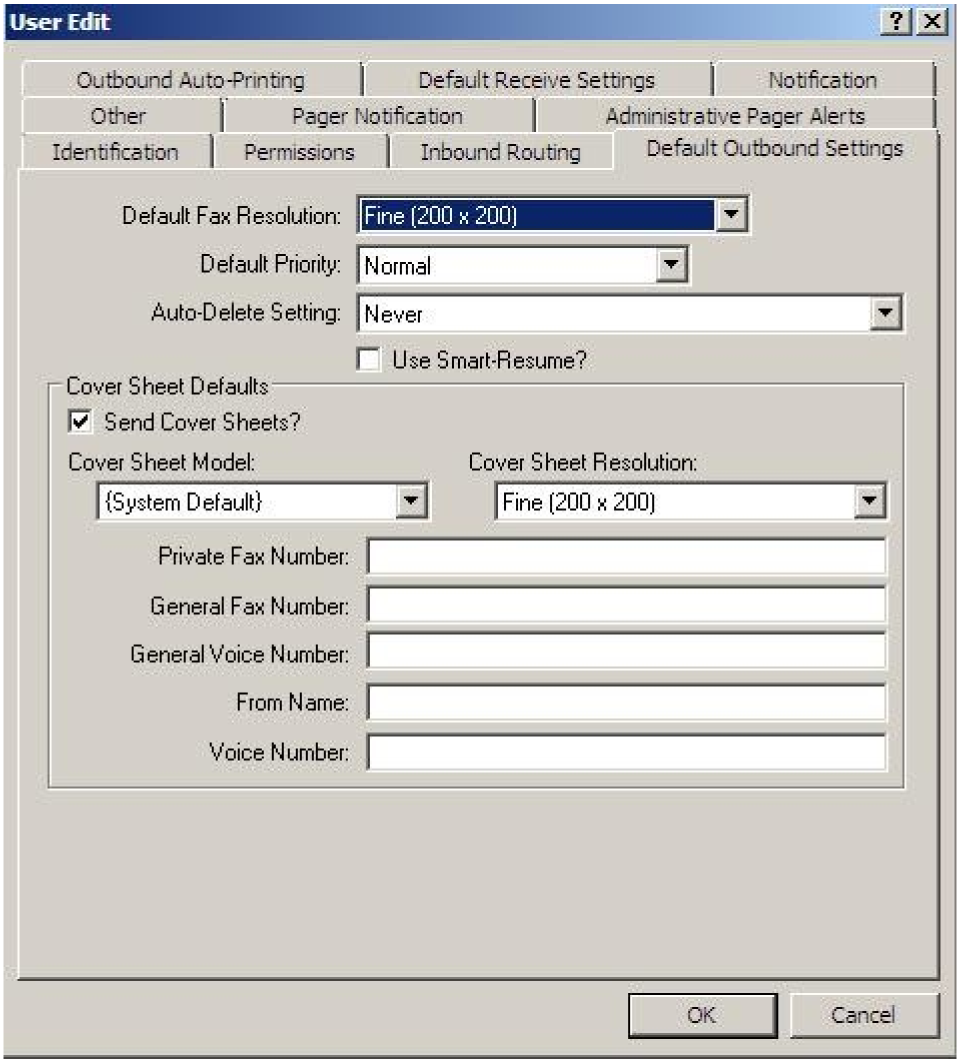
When routing to a Fax Mailbox, no additional information is necessary. If notifications occur through e-mail, the e-mail address should be specified in the Routing Info field.

Received Fax Routing Form: Advanced Outlook Form

☐ Delete after routing?

17. **Configure Users – Outbound Settings**

The **Default Outbound Settings** tab configures various outbound fax call settings. Configure these settings as appropriate.



Default Fax Resolution: Fine (200 x 200)
Default Priority: Normal
Auto-Delete Setting: Never
☐ Use Smart-Resume?

Cover Sheet Defaults
☑ Send Cover Sheets?
Cover Sheet Model: {System Default}
Cover Sheet Resolution: Fine (200 x 200)
Private Fax Number:
General Fax Number:
General Voice Number:
From Name:
Voice Number:

7. General Test Approach and Test Results

This section describes the compliance test approach used to verify interoperability of Open Text Fax Server, RightFax Edition with Avaya Aura® Session Manager.

7.1. General Test Approach

The general test approach was to make intra-site and inter-site fax calls to and from RightFax. The inter-site calls were made using SIP or ISDN-PRI trunks between the sites. Faxes were sent with various page lengths, resolutions, and at various fax data speeds. For capacity, a large number of 2-page faxes were continuously sent between the two RightFax servers simultaneously. Serviceability testing included verifying proper operation/recovery from failed cables, unavailable resources, and Session Manager and RightFax restarts. Fax calls were also tested with different Avaya Media Gateway media resources used to process the fax data between sites. This included the TN2302 MedPro circuit pack, the TN2602 MedPro circuit pack in the Avaya G650 Media Gateway; the integrated VoIP engine of the Avaya G450 Media Gateway and the Avaya MM760 Media Module installed in the Avaya G450 Media Gateway.

7.2. Test Results

Open Text Fax Server, RightFax Edition successfully passed all compliance testing.

7.3. General Observations

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 need to be installed in the Avaya G650 Gateway, and additional Avaya MM760 Media Module or Modules 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, G450, MM760	64	4
TN2602	64	1

Note that the SIP trunk group on Communication Manager for connecting to Session Manager at each site, as well as the SIP or ISDN-PRI trunk group for connecting the 2 sites must be configured with adequate number of trunk group members to support the number of simultaneous fax calls intended. On RightFax, adequate number of fax channels must also be appropriately configured for the intended capacity.

8. Verification Steps

The following steps may be used to verify the configuration:

- From Communication Manager SAT, use the **status signaling-group** command to verify that the SIP signaling groups configured in **Step 11** of **Section 4** are in-service.
- From Communication Manager SAT, use the **status trunk-group** command to verify that the SIP trunk group configured in **Section 4**, **Steps 12 - 13** is in-service.
- Verify that fax calls can be placed to/from Open Text Fax Server, RightFax Edition server at each site.
- From Communication Manager SAT, use the **list trace tac** command to verify that fax calls are routed to the expected trunks.
- From System Manager, confirm that the Entity Link between Session Manager and the Open Text Fax Server, RightFax Edition server is in service.

9. Conclusion

These Application Notes describe the procedures required to configure Open Text Fax Server, RightFax Edition server to interoperate with Avaya Aura® Communication Manager and Avaya Aura® Session Manager. Open Text Fax Server, RightFax Edition successfully passed compliance testing.

10. Additional References

[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 6.0, Issue 3, August, 2010.

[4] *Open Text Fax Server, RightFax Edition Administrator's Guide, Version 9.4 Feature Pack 1,* January 8, 2010.

[5] *Open Text Fax Server, RightFax Edition Installation Guide Version 9.4 Updated for Feature Pack 1 Service Release 3,* November 11, 2010.

Documentation for:

Avaya products may be found at http://support.avaya.com.
RightFax may be found at https://knowledge.opentext.com/knowledge/llisapi.dll/open/16512673.

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.